CALCULATIONS OF THE EFFECTS OF PEAK CLIPPING ON SPEECH-LIKE SIGNALS

WILLIAM VARNER DIETRICH

CALCULATIONS OF THE EFFECTS OF PEAK CLIPPING ON SPEECH-LIKE SIGNALS

by

William Varner Dietrich
Lieutenant, United States Navy
B.S., University of Wisconsin, 1962

Submitted in partial fulfillment
for the degree of

MASTER OF SCIENCE IN ENGINEERING ELECTRONICS

from the

NAVAL POSTGRADUATE SCHOOL
December 1966

ABSTRACT

Peak clipping is a well known method of increasing the average power output of a peak power limited voice communication transmitter. Although the clipping process introduces distortion, articulation tests have shown that clipped speech remains highly intelligible. Using idealizations of vowel sounds based on the mechanism of speech production, calculations were made of the spectra resulting from clipping these speech-like signals. The results indicate a high degree of similarity between the spectra before and after clipping. The power gained by clipping at audio frequency and at narrowband was calculated and compared with previously published data. Repeaking due to component rejection was investigated for clipping at audio and narrowband. Calculations of the effect of varying the phase characteristic of the signals before clipping indicate that such variation may improve the intelligibility of clipped speech.

TABLE OF CONTENTS

LIST OF TABLES

LIST OF ILLUSTRATIONS

1. Introduction

Although modern technology has provided the means to transmit large amounts of information efficiently, conventional voice communication systems have retained wide usage. While they are comparatively very inefficient, conventional speech communication systems will probably remain attractive for many applications because of their simplicity. An important factor contributing to the inefficiency of voice systems is the unsuitability of the speech signal for transmission by conventional means -- amplitude modulation or single sideband systems.

Speech waveforms characteristically have large peaks relative to their rms value. The peak factor, defined as

$$PF = 20 \log \frac{\text{peak value}}{\text{rms value}} ,$$

of speech is typically about 14.5 db. [14] In a conventional peak power limited transmitter, the amplitude of the modulated signal must be constrained so that the transmitter is not over-driven on the signal peaks. The average power in the speech signal being transmitted, then, is only a small fraction of the transmitter's peak power capability. For example, in a system where the information-bearing power is limited to 100 watts, a speech signal having a 14.5 db peak factor would have an average power of only 3.65 watts. Work done in speech processing for conventional transmission, therefore, has been focussed on methods of reducing the peak factor of speech waveforms.

The method most widely used is perhaps the most straightforward one: merely limit, or clip, the peaks of the speech signal. The clipping process, to be discussed further in Section 6, produces distortion. As the degree of clipping is increased the peak factor is reduced, but the amount of distortion generated increases. The trade-off between reduction of the peak factor and the increase in distortion is difficult to optimize, because it is not known how these two effects interact to influence the intelligibility of speech in noise.

The succeeding sections contain a brief description of the nature of speech and an electrical vocal tract analog. Following this is a discussion of some of the factors affecting speech intelligibility and a summary of the results of experiments conducted to determine the intelligibility of clipped speech.

In Section 7 a model of a speech signal is proposed for the purpose of calculating the spectrum which results when this speech-like signal is clipped. The computed spectra are presented and the reduction in the peak factor obtained is compared with previously published results. Repeaking of the clipped signals due to filtering and frequency translation is computed in Section 8. Finally, the effects of varying the phase characteristics of the speech-like signals on the clipping process are analyzed.

2. The Nature of Speech

A speech sound begins with a flow of air under pressure from the lungs through the trachea to the larynx. At the larynx the air stream may be periodically interrupted by the vibration of the vocal cords to produce a voiced sound, or it may pass through uninterrupted producing an unvoiced sound. In the case of a voiced sound, the periodic pulses of air produced by the vibration of the vocal cords excite the resonant modes of the vocal tract cavities to produce a distinctive sound. As the configuration of the vocal tract is changed, (by movement of the throat, tongue, and jaw, etc.) its acoustic transfer function changes so that different sounds are produced.

Unvoiced sounds are generated by forcing the air stream through constricted openings such as between the tongue and the upper teeth. These noise-like sounds are also modified by the resonances of the vocal tract. The mechanism for producing speech sounds is shown in block diagram form in Figure 1. [6]

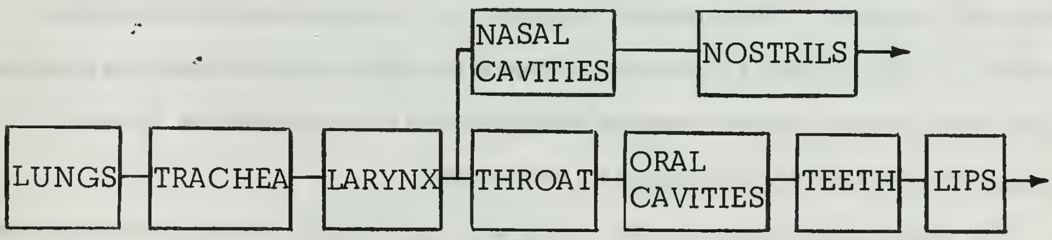


Fig. 1 Block diagram of voice mechanism

During normal breathing the vocal cords, which are actually two thick folds or bands, are widely separated at one end, forming a large triangular opening. To produce a voiced sound the cords are drawn together, closing off the air passage except for a thin slit. The act of exhaling then sets the vocal cords vibrating; the slit opens and closes periodically. The fundamental frequency of vibration varies from about 90 Hz for a deep voiced man to a maximum of about 350 Hz for a high-voiced woman, with typical values of 125 Hz and 250 Hz respectively.[6]

The vibration is by no means purely sinusoidal since the slit may remain closed for as long as half of the cycle. The sound spectrum produced therefore contains a fundamental and many harmonics. R. L. Miller has shown that the relative amount of energy present in the higher harmonics is related to the abruptness of closing the vocal cord slit. A speaker may increase the amplitudes of the higher harmonics by causing the slit to close more abruptly. [12] Speech sounds may have significant energy resulting from vocal cord action at frequencies as high as 40 times the fundamental frequency. [6]

The fundamental frequency, or pitch, of a voiced sound is changed by varying the tension of the vocal cords. As tension is increased the cords become firmer, and they are stretched to a greater length. The result is an increase in fundamental frequency.

Voiced sounds derive their distinguishing characteristics from the configuration of the vocal tract, the throat and oral cavities. Corresponding to a given configuration of the vocal tract there is a certain set of acoustic resonances which selectively transmit the harmonics of the sound source. The spectral regions of reinforcement are called formants. The action of the vocal tract in reinforcing some frequencies while attenuating others may be thought of as impressing a type of modulation on the source signal, or carrier.

The nasal tract acts in the same way to modify the source spectrum, but its configuration is essentially fixed.

3. Vocal Tract Analog

The mechanism for speech production, composed of an active part -- the sound sources -- and a passive part -- the vocal and nasal tracts, has led to the development of an electrical analog of the vocal tract. The vocal tract is viewed as an acoustic tube of varying cross-sectional area. If the cross-sectional dimensions are small compared to a wavelength of the sound, no error will be made by considering the tube to have a circular cross-section. The electrical analog of a cylindrical tube is a transmission line where current is analogous to volume velocity and voltage is analogous to sound pressure. [22] The characteristic impedance at any point on the transmission line analog depends on the cross-sectional area at the corresponding point on the acoustic tube. The distributed parameter transmission line is approximated by a series of lumped parameter sections where each section represents a given length, l, of the distributed line. The approximation is valid at frequencies where l is small compared to a wavelength. This condition, as well as the one concerned with cross-sectional dimensions, is satisfied at frequencies below 5 kHz. [22] The lumped parameter sections may be realized in the form of Γ, Π or T sections as shown in Figure 2.

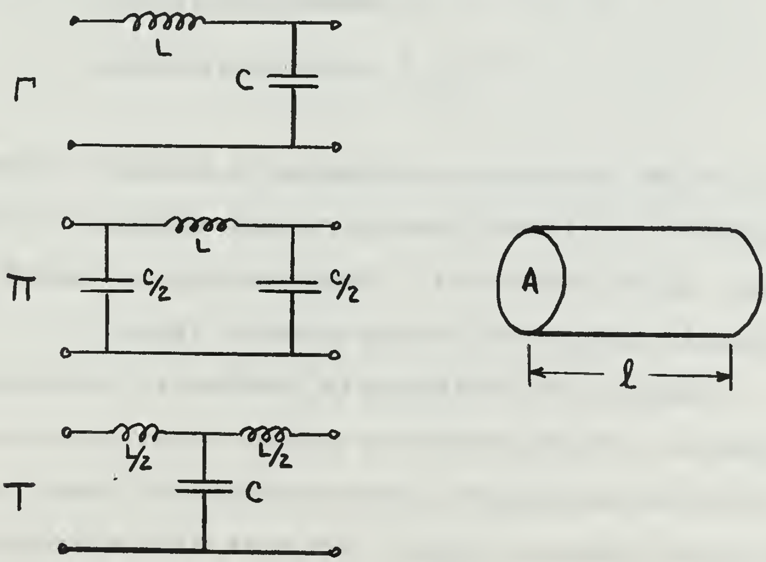


Fig. 2 Electrical Analogs of a short length of lossless acoustic tube.

The relations for the electrical parameters in terms of the dimensions of the acoustic tube are [23]:

$$L = \frac{\rho l}{kA} \qquad C = \frac{kAl}{\rho c^2}$$

where:

A = cross-sectional area of the tube

l = length of section

c = speed of sound

ρ = density of air

k = an arbitrary constant, the ratio of acoustic to electrical impedance

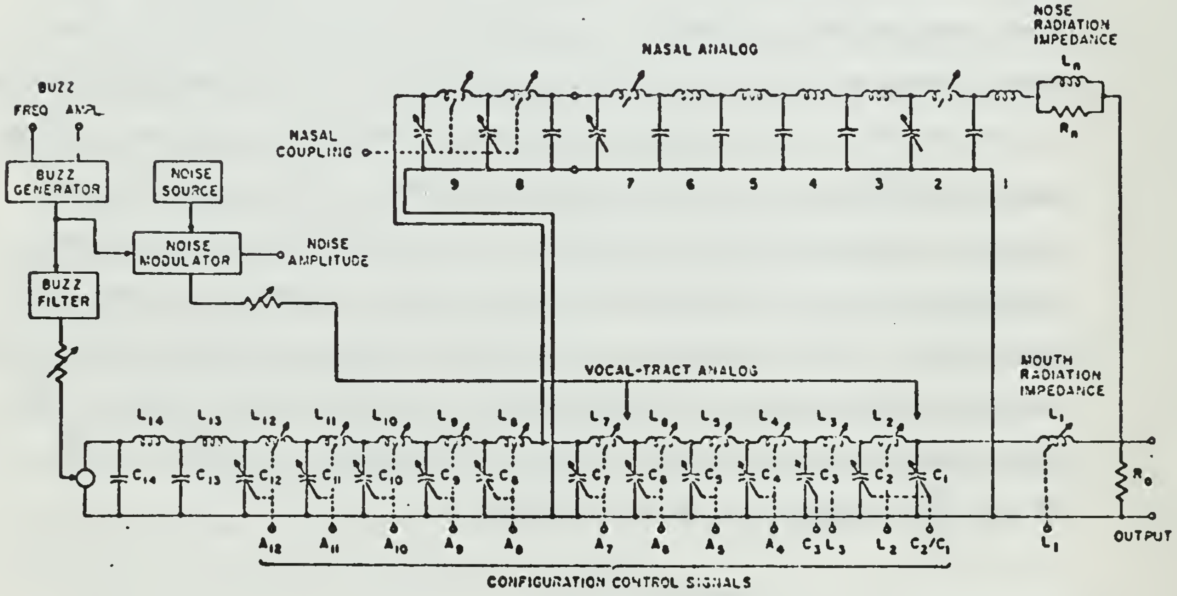


Fig. 3 Articulatory analog.

The lumped sections are connected in cascade to form an analog of an idealized vocal tract made up of approximately 35 cylinders, each $\frac{1}{2}$ cm long, placed end to end. When excitation is added the result is a complete articulatory analog shown in Figure 3.

Voiced sound excitation is provided in the analog by a "buzz generator" which produces a periodic train of pulses, narrow enough so that the amplitudes of the harmonics are nearly constant over the audio frequency range. The buzz filter attenuates the excitation so that the spectrum decreases at six db per octave with increasing frequency. The additional high frequency de-emphasis due to the output connection into the vocal tract analog produces a spectrum envelope that decreases at 12 db per octave.

For unvoiced sounds, excitation is provided by a gaussian noise source having a uniform spectrum extending from 100 to 10,000 Hz. Davenport has shown experimentally that the Gaussian distribution is a close approximation to the amplitude distribution of unvoiced speech sounds. [3] The noise source may be inserted at various points along the vocal tract analog according to the requirements of the sound being represented.

Speech synthesizers based on this type of an articulatory analog have produced good quality synthetic speech. Early models were most successful in producing vowel sounds, but more recent synthesizers, employing digital computers to control the vocal tract parameters, have been able to generate almost all of the elementary speech sounds as well as whole sentences. [23]

4. Speech Spectra and Intelligibility

The mechanism of speech production has naturally led researchers to characterize speech sounds by their frequency spectra. Voiced sounds, especially the vowels, are characterized by the location of their formants. Although the formant frequencies for a given vowel may vary from one speaker to another, their relative positions are similar. The principal formants, typically three in number, almost always occur at frequencies below 4 kHz. Figure 4 shows an idealized spectrum of the vowel sound in the word "bed".

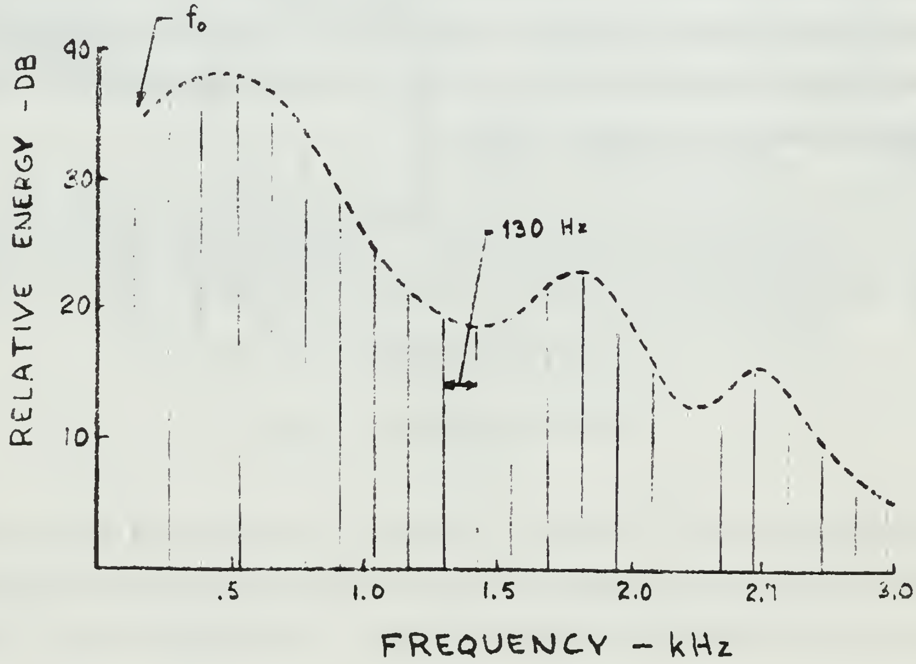


Fig. 4 Idealized spectrum of the vowel sound in "bed".

Unvoiced sounds have much less energy than voiced sounds due to the absence of vocal chord vibration. Their energy is spread over a broader frequency range, often extending to 8 kHz, and a distinctive formant structure is usually not present.

Researchers have concluded that the intelligibility of speech will remain good if the general shape of the power spectrum is preserved. This view has been confirmed by the success of vocoders in transmitting intelligible speech. The vocoder (for VOice CODER) represents an attempt to reduce the information rate, or bandwidth, necessary to transmit speech. Although there are many different types of vocoders, they are all basically analysis-synthesis schemes.

In the channel vocoder the analyzer takes the form of a bank of band-pass filters whose outputs, when rectified and smoothed, represent the envelope of the short-time spectrum. When the voiced-unvoiced detector indicates a voiced sound, the pitch extractor generates a signal proportional to the fundamental voice frequency.

The synthesizer is very similar in function to the articulatory analog discussed earlier. The filter signals are used to control the frequency response of a time-varying filter, realized as a bank of modulators followed by band-pass filters, so that the output spectral envelope corresponds to the one measured by the analyzer. The filter is excited by either a pulse generator for voiced sounds, or a white noise generator for unvoiced sounds. The frequency of the pulse generator is controlled by the pitch signal from the analyzer.

Other direct evidence that speech intelligibility is related to the preservation of the power spectrum comes from a system which measures an index of correlation between the patterns of the running power spectra of a distorted signal and the original undistorted speech signal. [20] The "pattern correspondence index" was found to have a direct relationship, by means of a calibration curve, to articulation test scores for various types of distortion, including peak clipping.

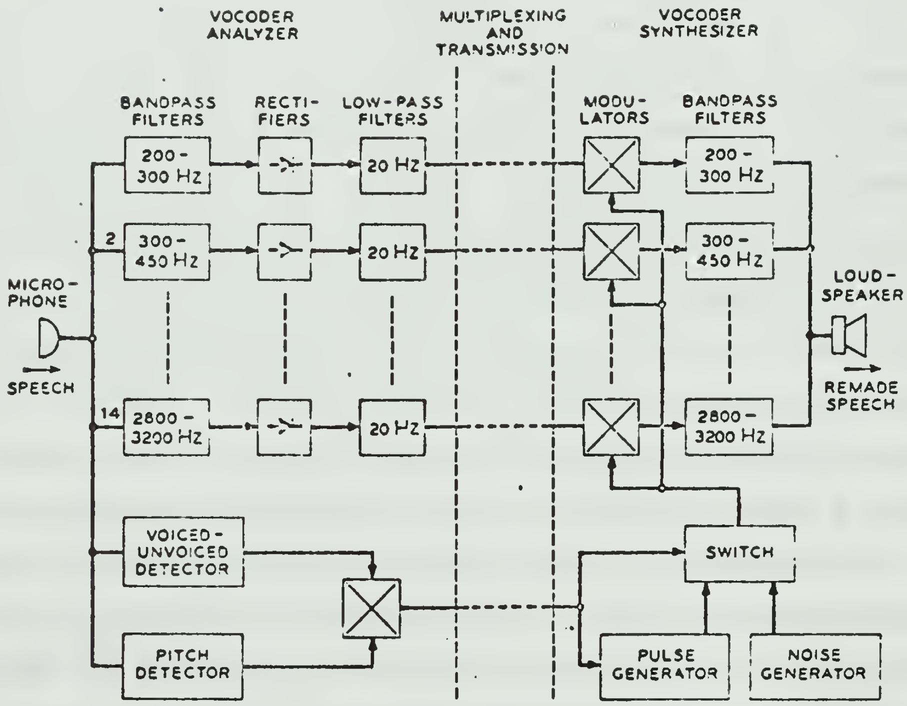


Fig. 5 Block diagram of a channel vocoder. [19]

5. Intelligibility of Clipped Speech

Studies of the intelligibility of clipped speech began near the end of World War II as a result of experiments conducted to determine the effects of overload distortion on intelligibility. It was found that speech remained at least moderately intelligible, although the quality suffered, no matter how much amplitude limitation was introduced.

An experiment conducted at Harvard University in 1944 under the direction of J.C.R. Licklider indicated that the intelligibility of speech in ambient noise remained essentially constant at a word articulation score of 80 percent as the amount of peak clipping was varied from 0 to 18 db. With 20 to 22 db clipping, articulation scores dropped to about 70 percent. [13] The subjective observations on the quality of the clipped speech noted were: a "sharp" or "scraping" effect due to the high frequency components generated in the clipping process and a monotonously uniform intensity. In addition it was found that with large amounts of clipping, noise picked up by the microphone was quite intense during the gaps between words.

Experiments at low listening levels near the threshold of audibility indicated that with clipped speech, reception was more uniform than with undistorted speech. Without clipping, intense words were audible but weaker ones were missed, but with clipped speech all of the words became audible at the same level. A closely related effect was the more nearly uniform audibility of the component sounds of test words under conditions of severe clipping. When undistorted speech was barely audible, the listener heard only the vowel sounds. However, when severely clipped speech was heard near the threshold of audibility, the consonants were as audible as the vowels. [13] The Harvard report found that the quality of speech was reduced "surprisingly little" by peak clipping. The experimenters observed that:

1) 6 db peak clipping is barely detectable;

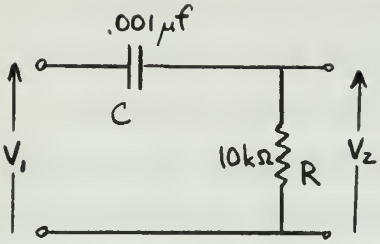
2) 12 db peak clipping is not at all objectionable, but, on the contrary, sounds as though the speaker were enunciating with special care;

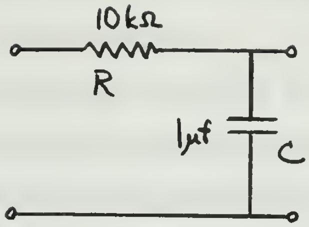
3) 18 db peak clipping makes speech sound somewhat sharp and rasping but less unnatural than speech over a throat microphone;

4) 24 db clipping leaves speech quite intelligible but makes it sound unnatural and "grainy". [13]

In a following study, Licklider and I. Pollack investigated the effects of differentiation and integration in combination with infinite clipping on speech intelligibility without noise. [10] (Infinite clipping implies that the speech signal is amplified by a very large factor prior to being clipped so that the output of the clipper is a series of positive and negative rectangular pulses of equal amplitude. The only characteristics retained from the original waveform are the times of zero crossings.) Integration and differentiation were accomplished by the circuits shown in Figure 6. The circuits were driven by low impedaace sources into high impedance loads. Over the frequency range of interest in speech, these circuits act as "spectrum tilters". As can be seen from the magnitude of the voltage transfer function, the differentiator (Figure 6a) attenuates the signal less as frequency increases; it tilts the spectrum upward at a rate of 6 db per octave. In similar fashion, the integrator tilts the spectrum downward, introducing an additional 6 db attenuation for each octave increase in frequency.

Articulation tests were run with ten different arrangements of integrator, differentiator, and infinite clipper. The results are plotted in Figure 7. The articulation scores fall into four groups. The first group is made up of the three tests where no clipping was done. Although the intelligibility scores for all three tests were near 100 percent, the observations on quality were very different. Differentiation, because it emphasizes the higher frequencies, made the speech sound overly crisp. On the other hand, integration made it sound muffled and "boomy".





$$\frac{V_2}{V_1} = \frac{R}{R + \frac{1}{j\omega C}} = \frac{j\omega}{10^5 + j\omega}$$

$$\left|\frac{V_2}{V_1}\right| \simeq \frac{\omega}{10^5}, \quad \omega < 10^4$$

a) differentiator

$$\frac{V_2}{V_1} = \frac{\frac{1}{j\omega C}}{R + \frac{1}{j\omega C}} = \frac{1}{1 + j\frac{\omega}{100}}$$

$$\left|\frac{V_2}{V_1}\right| \simeq \frac{100}{\omega}, \quad \omega > 1000$$

b) integrator

Fig. 6 The differentiating and integrating circuits used in the articulation tests.

The second group of intelligibility scores comes from the two arrangements where clipping was preceded by differentiation. The two sets of scores are almost the same, and both are always over 90 percent. Integration after clipping, while it did not affect intelligibility, did improve the quality.

The third group consists of the three cases where infinite clipping was the initial distortion: clipping alone, clipping plus differentiation, and clipping plus integration. In this group again it was found that the process following clipping had little effect on intelligibility. While integration after clipping again improved the quality, differentiation made the clipped speech sound even worse.

The final group of curves is the pair for which integration preceded infinite clipping. The articulation scores for this group are so

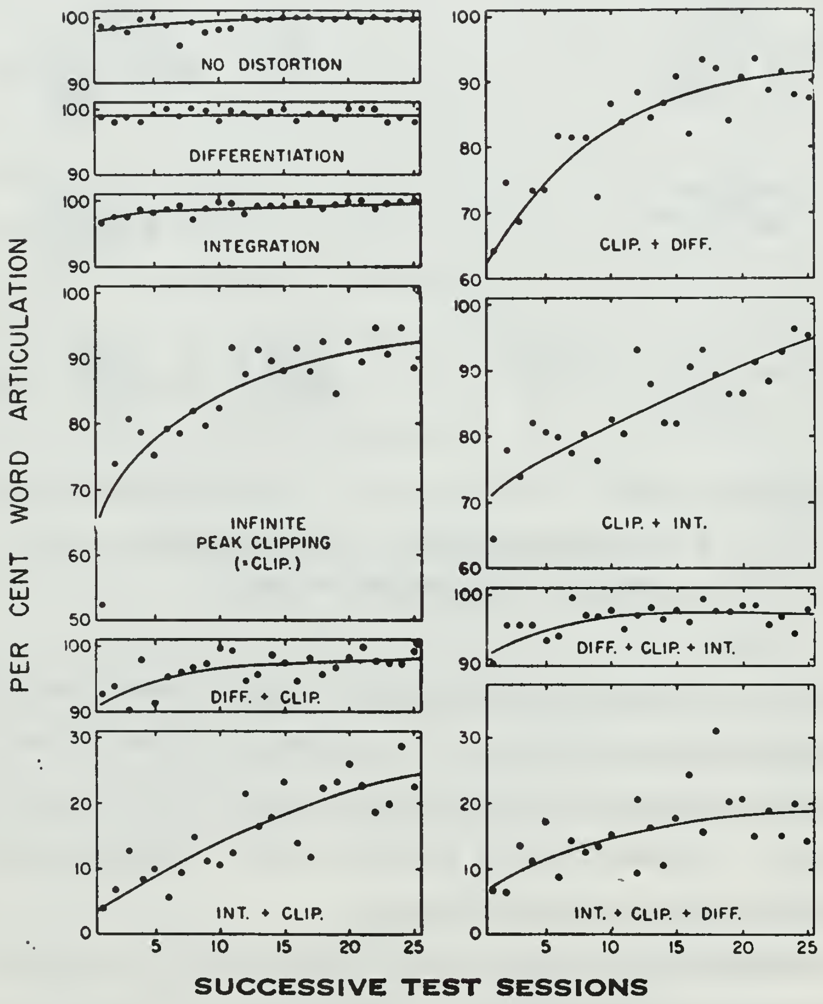


Fig. 7 Articulation scores for each of the ten arrangements of the distorting circuits. [10]

low that attenuating the higher frequencies and then clipping evidently distroys some of the important cues for intelligibility.

Additional work was done to try to determine why the intelligibility scores showed such marked improvement during the course of the experiment. Articulation tests were run using unfamiliar words with distortions of clipping and clipping plus differentiation. Scores on the new tests were about ten percentage points lower than the corresponding scores using the familiar words. However, scores for the new words were 15 points higher than the average of the first five original test sessions. The researchers concluded that, although familiarity with the test vocabulary (250 words) was an important factor, the listeners acquired a general ability to identify words correctly in spite of severe distortion.

A more recent study conducted at Montana State College in 1962 investigated the application of peak clipping to single sideband communications systems. [21] Intelligibility tests were run with signals which were clipped at various stages in the single sideband modulation process and then mixed with gaussian noise. The signal-to-noise ratio defined for these tests was:

$$\lambda = 20 \log_{10} \frac{E_p}{E_n}$$

where E_p = clipped signal peak value

E_n = rms valud of the noise

Signal and noise have the same bandwidth

This definition is particularly appropriate to evaluation of processing techniques for peak power limited systems.

Figure 8 shows the intelligibility scores resulting from audio clipping followed by low-pass filtering for a 5 kHz bandwidth. In this case, differentiation prior to clipping did not result in improved intelligibility. This may not, however, contradict Licklider and Pollack's data on clipping without noise. These results will be

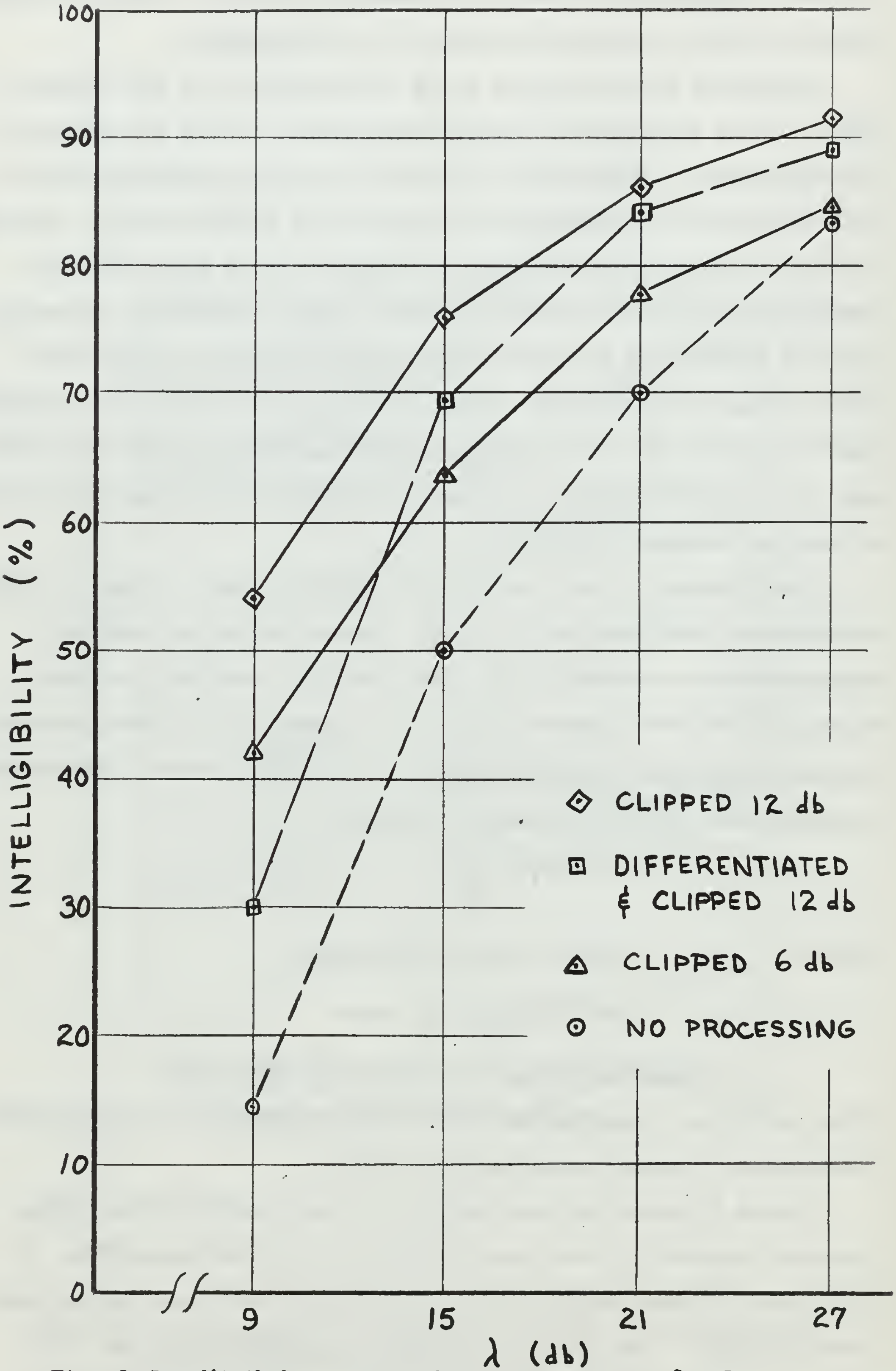


Fig. 8 Intelligibility scores for audio clipping [21].

discussed further in Section 7. Figure 9 shows the effect of differentiation prior to clipping the single sideband signal. These results give no clear indication that differentiation either reduces or enhances intelligibility.

The most significant results from the Montana State study are those comparing pre-modulation and post-modulation clipping for single sideband transmission. The scores plotted in Figure 10 show that, in every case, clipping the single sideband signal provided improvement in intelligibility over clipping the audio signal before modulation. For both cases, λ was defined at the output of the SSB filter.

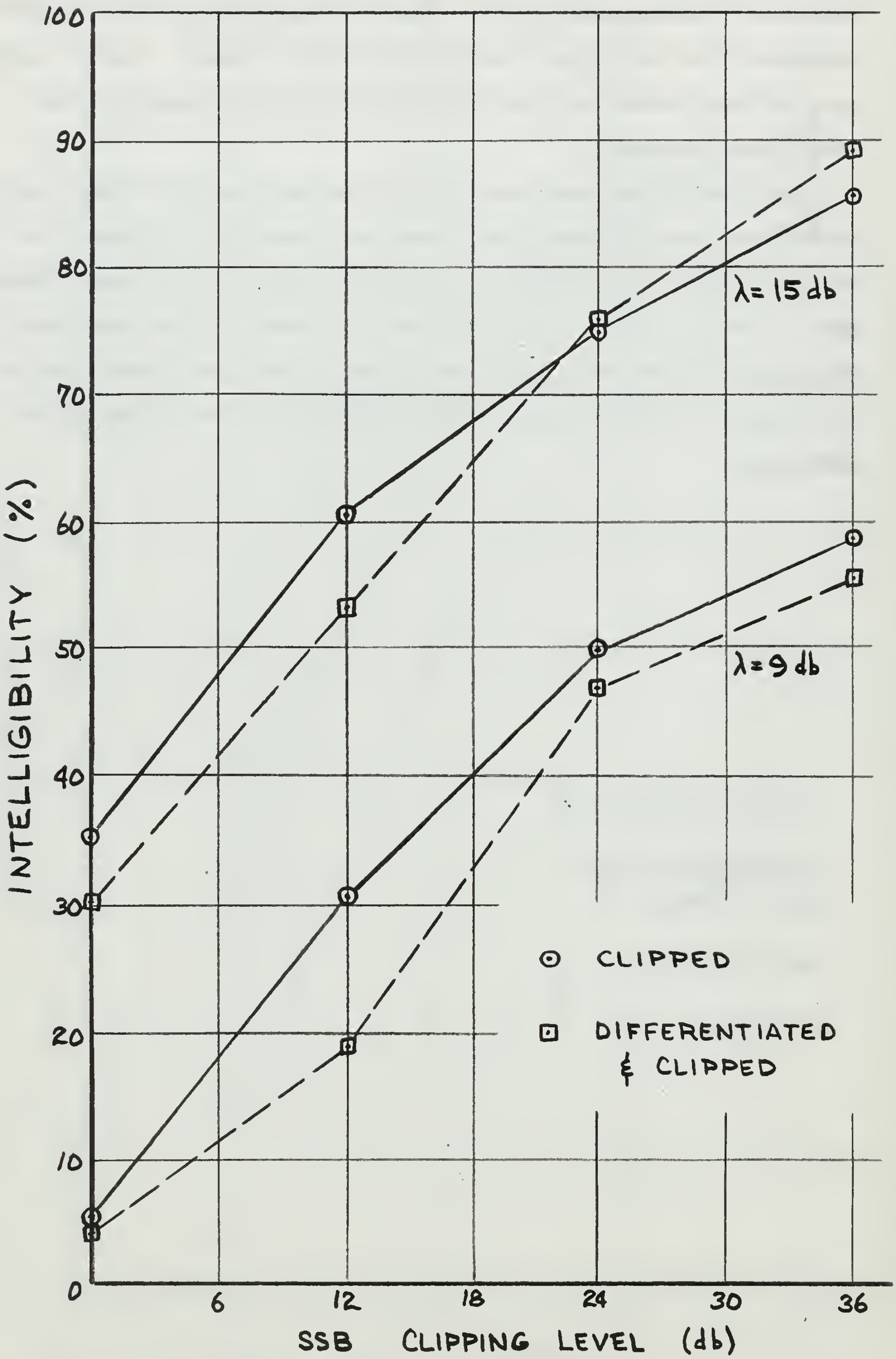


Fig. 9 Intelligibility with and without differentiation prior to SSB clipping [21].

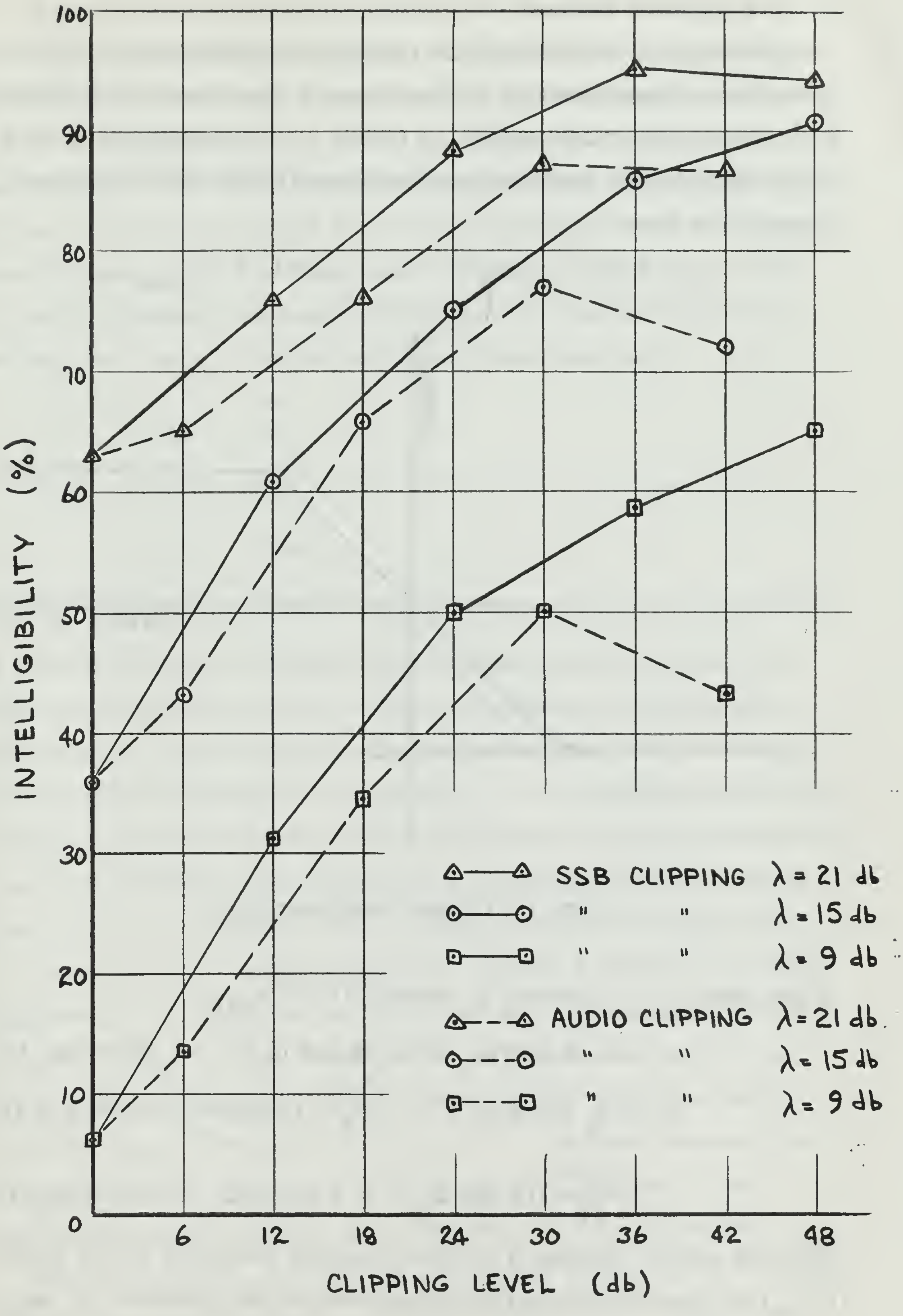


Fig. 10 Comparative intelligibility for pre-modulation and post-modulation clipping [21].

6. The Clipping Process

The clipping process may be represented mathematically by a power series approximation to the clipper's input-output characteristic. The clipper characteristic in Figure 11 is an odd function of x, therefore its power series approximation will have only odd-order terms of the form:

$$y = a_1 x + a_3 x^3 + a_5 x^5 + \ldots, \quad |x| \leq x_{max}$$

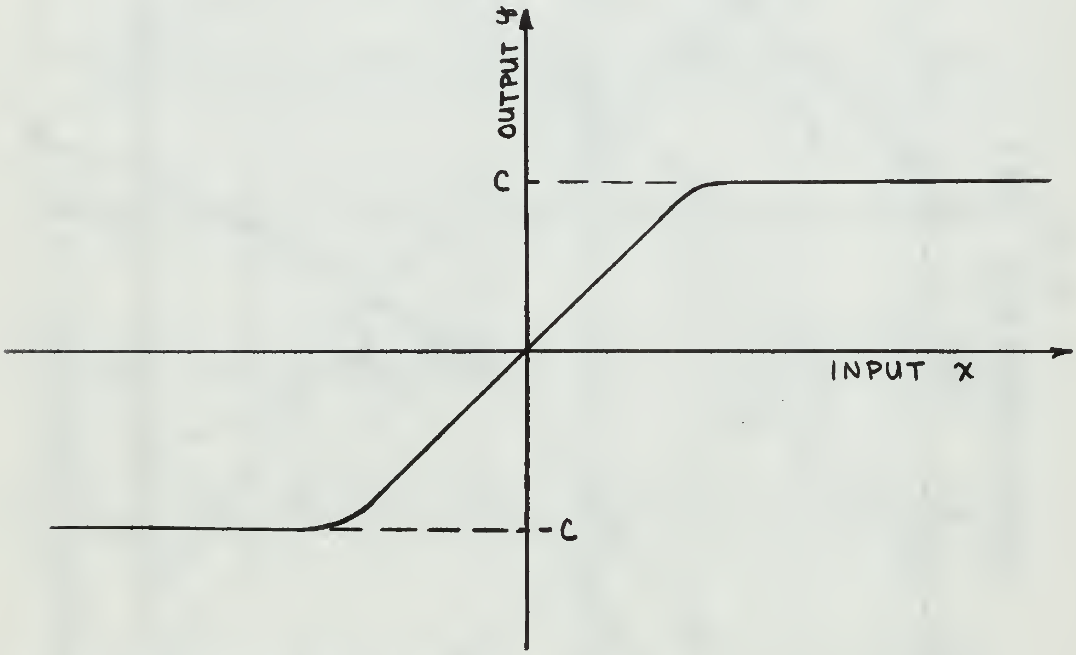


Fig. 11 Clipper characteristic

If the input $x = A \cos \omega_o t$ where $C < A < x_{max}$:

$$y = a_1 (A \cos \omega_o t) + a_3 (A \cos \omega_o t)^3 + a_5 (A \cos \omega_o t)^5 + \ldots$$

$$y = a_1 A \cos \omega_o t + \frac{a_3 A}{4} (3 \cos \omega_o t + \cos 3 \omega_o t) +$$

$$\frac{a_5 A}{16} (10 \cos \omega_o t + 5 \cos 3 \omega_o t + \cos 5 \omega_o t) + \ldots$$

Thus the output contains a component at the frequency of the input, ω_o, plus harmonic distortion components at odd multiples of ω_o .

If the input is a two-tone signal of the form $x = A \cos \omega_1 t + B \cos \omega_2 t$, the output will contain, in addition to the fundamentals and odd-order harmonics, intermodulation distortion components at frequencies given by $i\omega_1 \pm j\omega_2$, where i and j are positive integers and (i + j) is odd. Table I lists the fifth order approximations to the amplitudes of the spectral components. If the signal being clipped is a narrow-band signal, ($|\omega_1 - \omega_2| \ll \omega_1$) only four of the 16 distortion components listed will fall at frequencies near ω_1 and ω_2. The frequencies of these four are:

$$2\omega_1 - \omega_2$$

$$2\omega_2 - \omega_1$$

$$3\omega_1 - 2\omega_2$$

$$3\omega_2 - 2\omega_1$$

All other distortion components will be separated from these by at least the order of ω_1 rad/sec. If the approximation is extended to seventh order, there will be a total of 30 distortion components, of which only six will fall near ω_1 and ω_2. The remaining distortion components will lie outside the frequency band of interest and may be removed by filtering. This reduction of distortion helps explain why speech is more intelligible when clipped at narrow-band (SSB) than it is when clipped at audio frequency. Figure 12 shows the relative amplitudes of the spectrum resulting from infinitely clipping a signal of the form $\cos 2\pi m f_o t + \cos 2\pi (m+1) f_o t$. For an input consisting of two cosine waves, the spectrum of the clipped signal is of the form $A_n \cos 2\pi n f_o t$, n = 1, 2, ... (An even function clipped symmetrically remains an even function.) In Figure 12 the negative amplitudes indicate phase reversal of the components.

When m is small, the input is a wideband signal and the clipping process has a marked effect on the amplitudes of the "signal" frequencies -- mf_o and $(m + 1)f_o$. (For the case m = 1, the two

Table I

Amplitudes of spectral components resulting from clipping a signal of the form $A \cos \omega_1 t + B \cos \omega_2 t$.

FREQUENCY (rad/sec)	AMPLITUDE
ω_1	$a_1 A + 3/4\, a_3 A^3 + 3/2\, a_3 AB^2 + 5/8\, a_5 A^5 + 15/4\, a_5 A^3 B^2 + 15/8\, a_5 AB^4 + \ldots$
ω_2	$a_1 B + 3/4\, a_3 B^3 + 3/2\, a_3 A^2 B + 5/8\, a_5 B^5 + 15/4\, a_5 A^2 B^2 + 15/8\, a_5 A^4 B + \ldots$
$3\omega_1$	$1/4\, a_3 A^3 + 5/16\, a_5 A^5 + 5/4\, a_5 A^3 B^2 + \ldots$
$3\omega_2$	$1/4\, a_3 B^3 + 5/16\, a_5 B^5 + 5/4\, a_5 A^2 B^3 + \ldots$
$2\omega_1 \pm \omega_2$	$3/4\, a_3 A^2 B + 5/4\, a_5 A^4 B + 15/8\, a_5 A^2 B^3 + \ldots$
$\omega_1 \pm 2\omega_2$	$3/4\, a_3 AB^2 + 5/4\, a_5 AB^4 + 15/8\, a_5 A^3 B^2 + \ldots$
$5\omega_1$	$1/16\, a_5 A^5 + \ldots$
$5\omega_2$	$1/16\, a_5 B^5 + \ldots$
$4\omega_1 \pm \omega_2$	$5/16\, a_5 A^4 B + \ldots$
$3\omega_1 \pm 2\omega_2$	$5/8\, a_5 A^3 B^2 + \ldots$
$2\omega_1 \pm 3\omega_2$	$5/8\, a_5 A^2 B^3 + \ldots$
$\omega_1 \pm 4\omega_2$	$5/16\, a_5 AB^4 + \ldots$

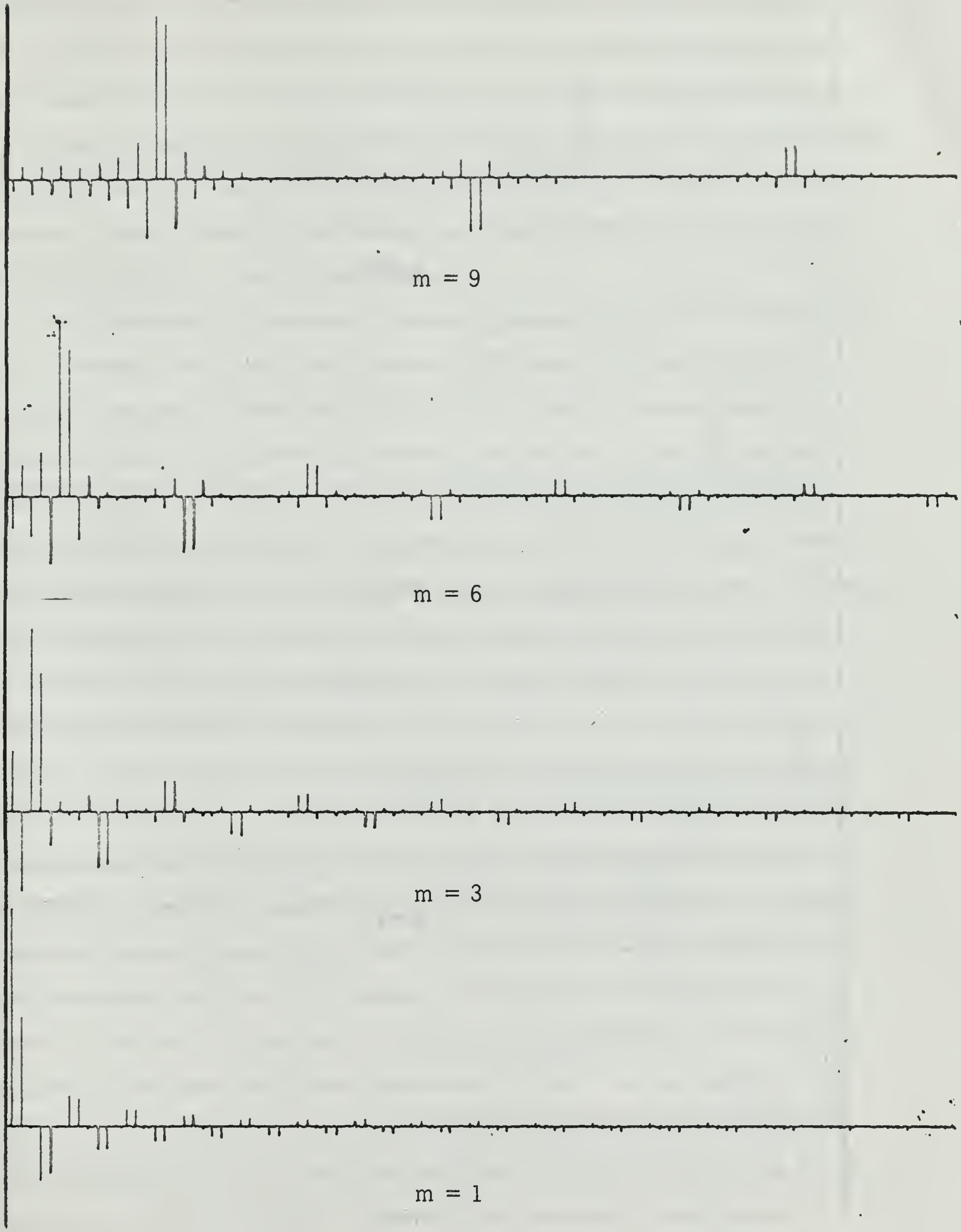


Fig. 12 Relative amplitudes of the spectral components resulting from infinitely clipping signals of the form $\cos 2\quad mf_o t + \cos 2\quad (m+1)f_o t$

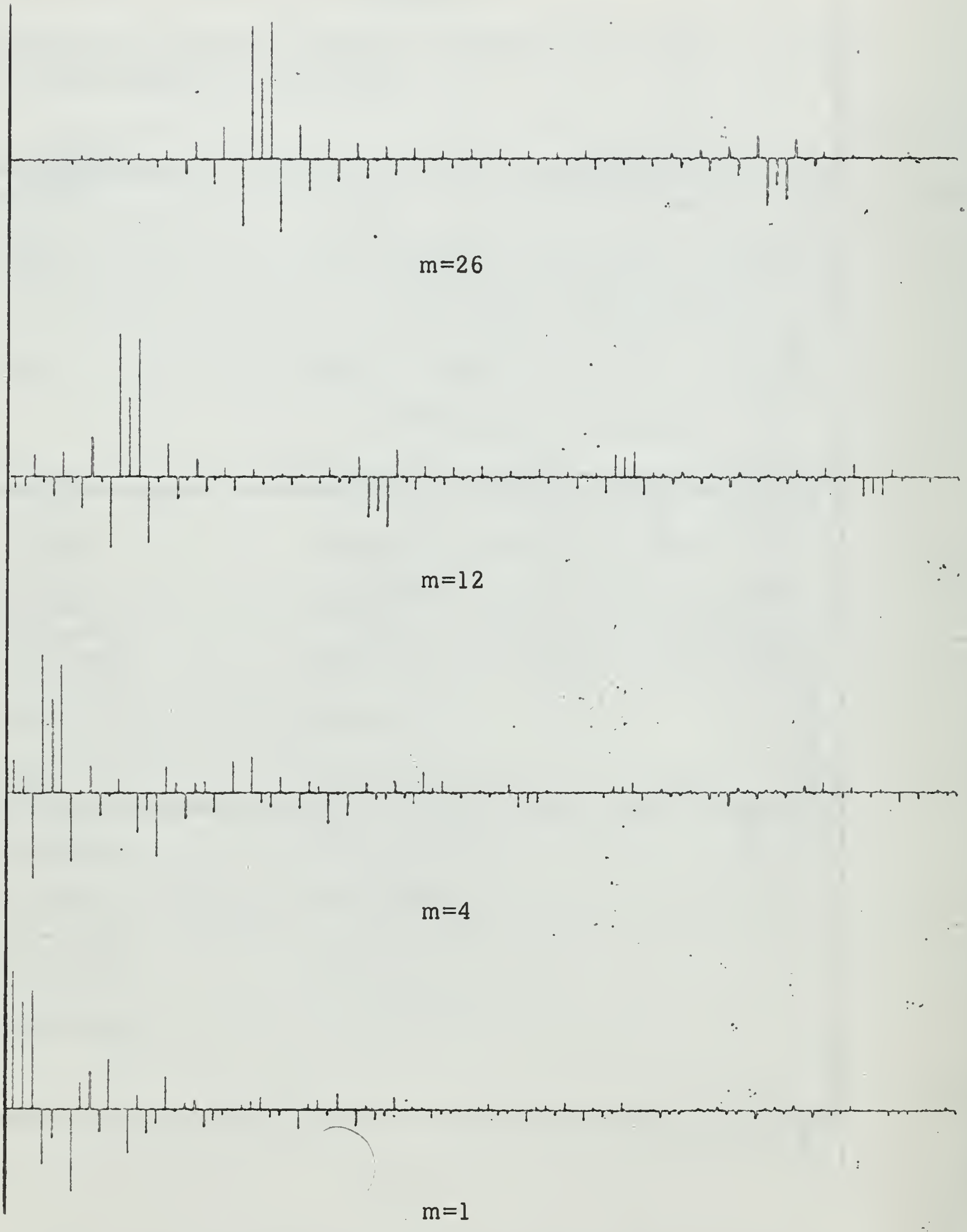


Fig. 13 Relative amplitudes of the spectral components resulting from infinitely clipping signals of the form $\cos 2\quad mf_o t + \cos 2\quad (m+1)f_o t + \cos 2\quad (m+2)f_o t$

originally equal amplitudes have been distorted so that the component at $2f_o$ is only half as large as the one at f_o.) As m increases, the input gradually changes from a wideband to a narrowband signal and a relatively simple spectral pattern emerges. The pattern repeats itself (with reversed sign) at odd multiples of $(m + \frac{1}{2})\, f_o$. Figure 13 shows the same type of result for a three-tone input of the form $\cos 2\pi m f_o + \cos 2\pi (m + 1) f_o + \cos 2\pi (m + 2) f_o$.

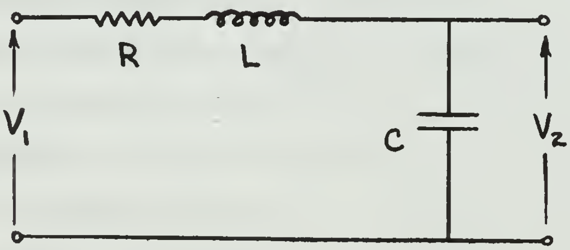
J. M. Dukes [4] analyzed the clipping process from a statistical point of view. He found that, for a totally random signal, infinite clipping does not change the shape of the signal's power spectral density function. For partially constrained signals such as voiced speech sounds, however, the shape of the power spectral density curve is altered by clipping. In connection with this finding, Dukes sites the results of an investigation which showed that vowels suffer more reduction in intelligibility due to clipping than do consonants.

Since the short-time spectrum of a vowel consists entirely of harmonically related components, i.e., $f_i = i f_o$, $i = 1, 2, 3, \ldots$, clipping will generate no new frequencies in the band of the original signal. The distortion components will fall either at the frequencies of the original spectral components or at higher frequencies where the original spectrum amplitudes were insignificantly small. Thus, distortion measurements made with simple sine waves or two-tone signals cannot indicate the degree to which the intelligibility of a speech signal will be degraded. Some of the distortion generated will lie at frequencies higher than those present in the original signal, and this portion might be considered to affect intelligibility in the same manner as corrupting noise. But much of the distortion will simply alter the amplitudes of the original spectral components.

7. Results of Calculations

In order to investigate the alteration of the spectral amplitudes due to peak clipping, calculations were made, with aid of a Control Data 1604 computer, of the spectra of clipped speech-like signals. The signals used were idealizations of vowel sounds described in the frequency domain by their spectrum amplitude characteristics. The signals were considered to be periodic with a period of 1/120 sec corresponding to a fundamental voicing frequency of 120 Hz. Thus the idealized spectrum of the phomeme / / in the word "bed" could be represented by the amplitudes of 23 spectral components covering frequencies up to 2760 Hz.

The relative phases of the components were computed by inferring a phase characteristic from the amplitude characteristic. Considering the vocal tract analog discussed in Section 3, a peak in the amplitude characteristic could be considered with small error to result from a complex conjugate pole-pair in the transfer function or, alternatively, from the resonance of a single L-C section. A discussion of the errors involved in this approximation is given in Reference 26.



Single section of vocal tract analog.

The voltage transfer function of the circuit shown above is:

$$\frac{V_2}{V_1} = \frac{1}{s^2LC + sRC + 1}$$

$$= \frac{1}{(1 - \omega^2 LC) + j\omega RC}$$

The phase of V_2 with respect to V_1 is:

$$\phi = -\tan^{-1}\left[\frac{\omega RC}{1-\omega^2 LC}\right]$$

$$= -\tan^{-1}\left[\frac{\omega RC}{1-\dfrac{\omega^2}{\omega_o^2}}\right]$$

$$= -\tan^{-1}\left[\frac{\omega_o RC}{\dfrac{\omega_o}{\omega}-\dfrac{\omega}{\omega_o}}\right]$$

but $\omega_o RC = \dfrac{1}{Q_o}$

then $$\phi = \tan^{-1}\frac{1}{Q_o\left(\dfrac{f}{f_o}-\dfrac{f_o}{f}\right)}$$

The phase characteristic corresponding to an amplitude function having, for example, three formants is then the superposition of three phase functions of the type shown above, one for each resonant frequency. The Q factor for each phase function is dependent upon the formant frequency and bandwidth according to the relation:

$$BW_{3db} = \frac{f_o}{Q_o}$$

For this investigation, formant bandwidths were set at 100 Hz, a mean value taken from data in Reference 1.

In Figure 14 the amplitude characteristic for the vowel sound in the word "bed" is plotted as a series of straight line segments connecting successive points representing the amplitudes of the 23 discrete frequency components. The curve labeled "PHI" is the phase characteristic inferred by the process previously discussed. The amplitude characteristic was normalized so that the corresponding time function would have a maximum absolute value of 10. The curve labeled "PWR" is, in effect, a power distribution curve: at any given frequency it represents the total power (across a one ohm resistor) in the components at that frequency and below. Figures 15, 16, and 17 show the spectra resulting from clipping the time waveform 10 db, 20 db, and infinitely. To get, for example, 10 db clipping the time function was amplified by a factor of 3.16 and then clipped symmetrically at +10 and -10. The clipping characteristic was ideally "abrupt" in that, for values of the input between +10 and -10, the clipper output was identical to the input, and for values exceeding that range the output was fixed at +10 or -10 as appropriate. In these Figures, the last point on the power distribution curve represents the total power in the clipped wave. For the infinitely clipped wave, the total power reaches the maximum possible value of 100.

Figure 18 shows the spectrum resulting from infinitely clipping not the audio waveform but the corresponding upper sideband signal. This signal was generated according to the equation

$$f_{USB}(t) = \sum_{n=1}^{N} C_n \cos[2\pi(f_c + nf_o)t + \phi_n]$$

where C_n represents the amplitude characteristic and $_n$ the phase characteristic. The carrier frequency, f_o, was chosen as 24 kHz, a value high enough to give meaningful results but low enough so that the number of computations required for fourier analysis would not be unreasonably large. The value of f_o remained at 120 Hz. To save time, the clipped signal was analyzed only at the frequencies of the 23 components present in the original signal.

Similar calculations were performed using models of three other vowel sounds, the phonemes /a/, /ɛ/ and /u/. The amplitude characteristics for these models were taken from spectrum envelopes

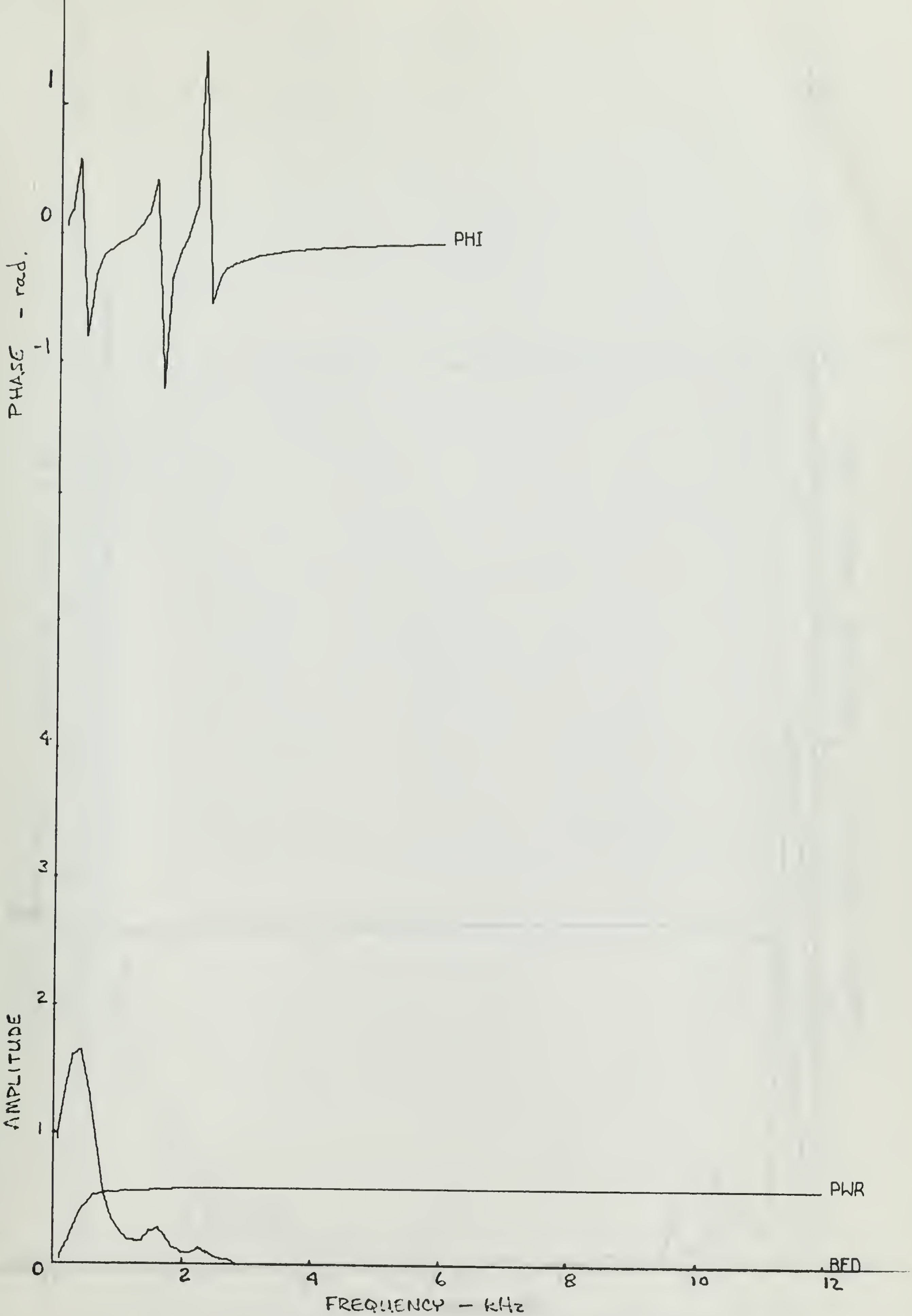


Fig. 14 Amplitude and phase characteristics for the vowel sound in "bed".

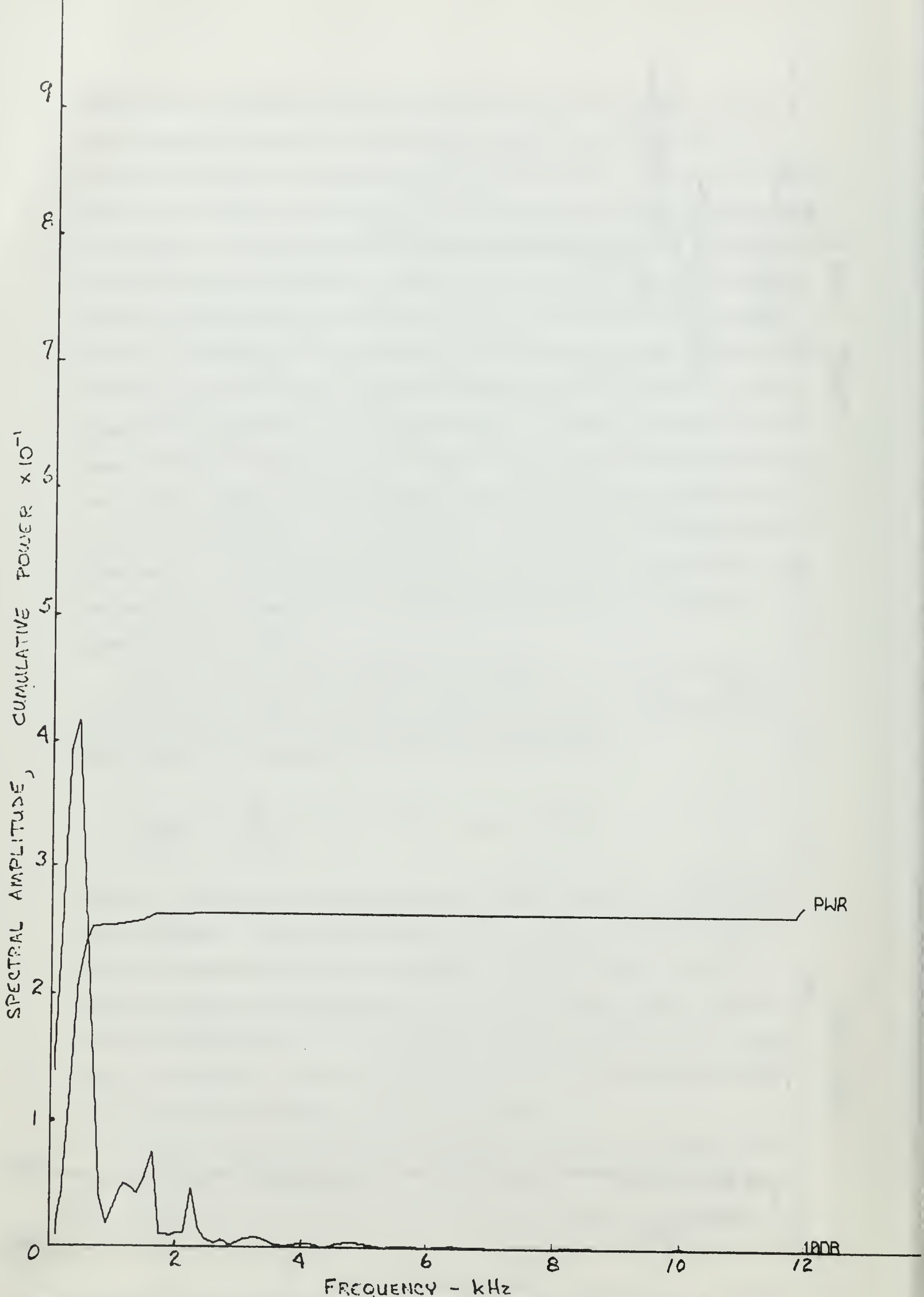


Fig. 15 Spectrum of "bed" clipped 10 db

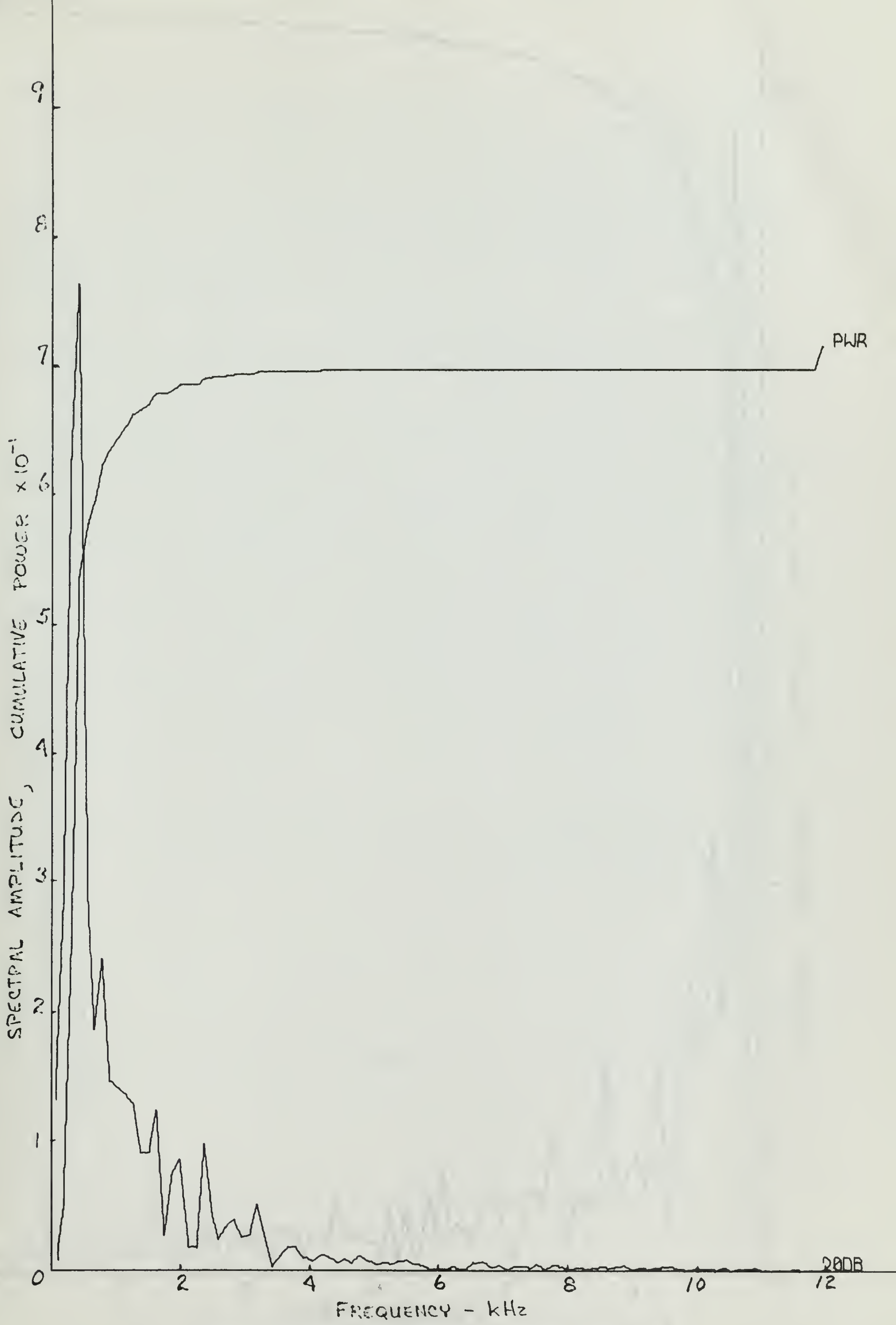


Fig. 16   Spectrum of "bed" clipped 20 db

Fig. 17 Spectrum of "bed" infinitely clipped

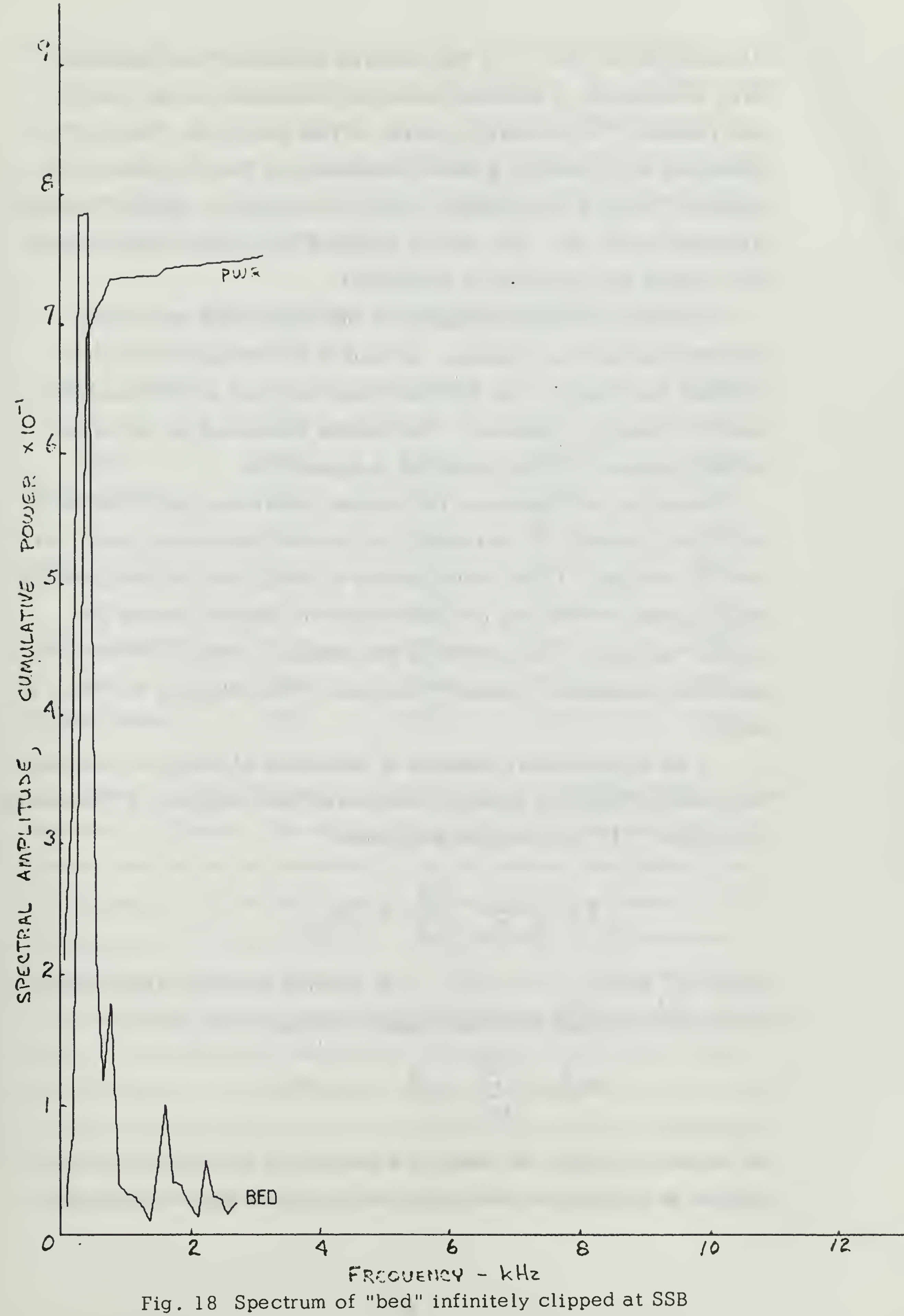


Fig. 18 Spectrum of "bed" infinitely clipped at SSB

calculated by G. Fant. [5] The spectrum envelopes were mathematically synthesized in terms of elementary resonance curves, one for each formant. The process is similar to that previously discussed in connection with inferring a phase characteristic from the formant frequencies. For his calculations, Fant also assumed a constant formant bandwidth of 100 Hz. The spectra resulting when these sound models were clipped are presented in Appendix A.

In addition the same calculations were made with the signals differentiated prior to clipping. Instead of differentiating the time waveform numerically, the differentiation was done in the frequency plane by tilting the spectrum. The spectra calculated for the differentiated signals are also presented in Appendix A.

Generally, the spectra of the clipped signals are quite similar to the orignal spectra. In particular, the formant frequencies are not altered by clipping. In all of the spectra of the clipped signals, smaller peaks appear between the formants where the original spectra had smooth "valleys." The spectra of the signals clipped at SSB are, as expected, noticeably "cleaner" than those where clipping was done at audio.

To get a quantitative measure of the degree of similarity between the shape of the power spectrum before and after clipping, a "coherence coefficient," γ, was defined as follows:

$$\gamma = \frac{1}{\sigma_1 \sigma_2} \sum_{i=1}^{N} C_{1i} C_{2i}$$

where C_{1i} and C_{2i}, $i = 1,2,\ldots,N$, are the discrete values of the power spectra before and after clipping, respectively, and

$$\sigma_i^2 = \sum_{i=1}^{N} C_{ji}^2 \quad , \quad j = 1,2$$

For each computation the value of N was set so that only those components at frequencies which were present in the original signal were included. The higher frequency distortion components generated in the clipping process did not enter into the calculation. In the cases where differentiation preceded clipping, the coherence coefficient was calculated using the spectrum of the differentiated signal as the original spectrum before clipping. The coherence coefficients calculated in the various tests are listed in Table II.

TABLE II

Coherence Coefficients After Clipping

model sound	at audio 10db	20db	inf.	at SSB inf.
"bed"	.9688	.8808	.8430	.8723
differentiated	.9669	.9046	.8536	.8506
/ /	.9826	.9747	.9725	.9694
differentiated	.9920	.9818	.9744	.9907
/a/	.9623	.9238	.9072	.9320
differentiated	.9816	.9647	.9551	.9604
/u/	.9920	.9815	.9805	.9941
differentiated	.9707	.9422	.9354	.9764

Listed in Table III are the values of power calculated after various amounts of clipping. The calculated power increases as clipping becomes more severe because the signal was always amplified so that its maximum absolute value was 10. The peak factor before clipping, averaged over the four signals, is 10.9 db. For the differentiated signals the average peak factor is 12.3 db. In every case the differentiated signal has less power than the corresponding non-differentiated wave. This result may explain why the Montana State study found differentiated and clipped speech to be less intelligible in noise whereas Licklider and Pollack found that differentiation before clipping enhanced intelligibility without noise. While, in almost all cases, the differentiated signals have larger coherence coefficients and therefore

TABLE III

Results of Power Calculations

Sound model	Clipping level	Audio Clipping: Power in band of original signal	Audio Clipping: Total power	SSB Clipping: Power in band-width of original signal
"bed"	0	5.9	5.9	
	10 db	26.6	27.3	
	20 db	69.3	71.5	
	inf	92.5	100.0	75.0
differen-tiated	0	3.8	3.8	
	10 db	17.4	18.1	
	20 db	55.3	59,3	
	inf	78.1	100.0	70.5
/ɛ/	0	7.7	7.7	
	10 db	46.4	47.1	
	20 db	78.3	83.6	
	inf	86.8	100.0	76.1
differen-tiated	0	7.0	7.0	
	10 db	40.8	42.1	
	20 db	67.7	79.0	
	inf	75.5	100.0	70.4
/a/	0	5.8	5.8	
	10 db	35.6	35.9	
	20 db	72.3	75.8	
	inf	85.9	100.0	77.4
differen-tiated	0	4.8	4.8	
	10 db	29.7	30.3	
	20 db	67.8	73.5	
	inf	83.1	100.0	75.6
/u/	0	17.3	17.3	
	10 db	68.1	69.0	
	20 db	92.0	93.2	
	inf	95.9	100.0	79.6
differen-tiated	0	9.7	9.7	
	10 db	47.9	48.2	
	20 db	76.0	77.6	
	inf	91.1	100.0	72.9

presumably greater intelligibility, perhaps the difference in power is the dominant factor when speech is masked by noise.

In Figure 19 the average power increase in the band of the original signal is plotted vs. the amount of audio clipping. The higher points represent the average power increase for the signals which were differentiated prior to clipping; the lower points are averaged over the non-differentiated signals. It should be noted that, for this discussion, power increase in db is equivalent to peak factor reduction in db since all signals have the same peak value.

The increase in power due to clipping was calculated in Reference 25 from a speech amplitude probability distribution. The result of this calculation is plotted in Figure 20 along with the average increase in total power of the four signals from this experiment. The other points on the graph represent data replotted from Figure 8 by taking the values of λ for which the curves for 12 db clipping, 6 db clipping, and no processing cross the 60% intelligibility line. For example, with no processing 60% intelligibility corresponds to a λ of about 18 db and with 6 db clipping the corresponding λ is 13.5 db. The difference between these two values of λ should be a close approximation to the reduction in peak factor due to 6 db clipping.

In Figure 21, the in-band power is plotted versus the coherence coefficient. Although the relationship appears to be nearly linear, the slopes of the curves vary widely. In the case of the model sound /a/, data for the differentiated signal has a larger slope than for the non-differentiated wave, but for /u/ the reverse is true. For the sake of clarity, points were not plotted for the signals clipped at SSB, however these data have no consistent relationship to the curves shown.

N. W. Huddy [8] investigated the effects on intelligibility of using a clipper having a gradual clipping characteristic instead of the more common abrupt clipper. It was thought that the gradual clipper might give better results since its input-output characteristic did not have a sharp bend. At modest clipping levels, the output waveform would

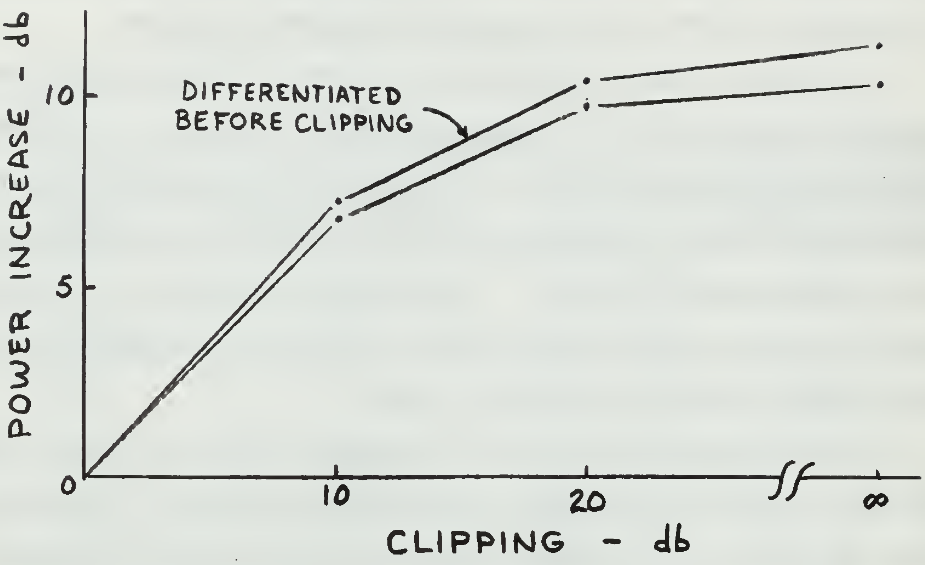


Fig. 19 Average in-band power increase with clipping

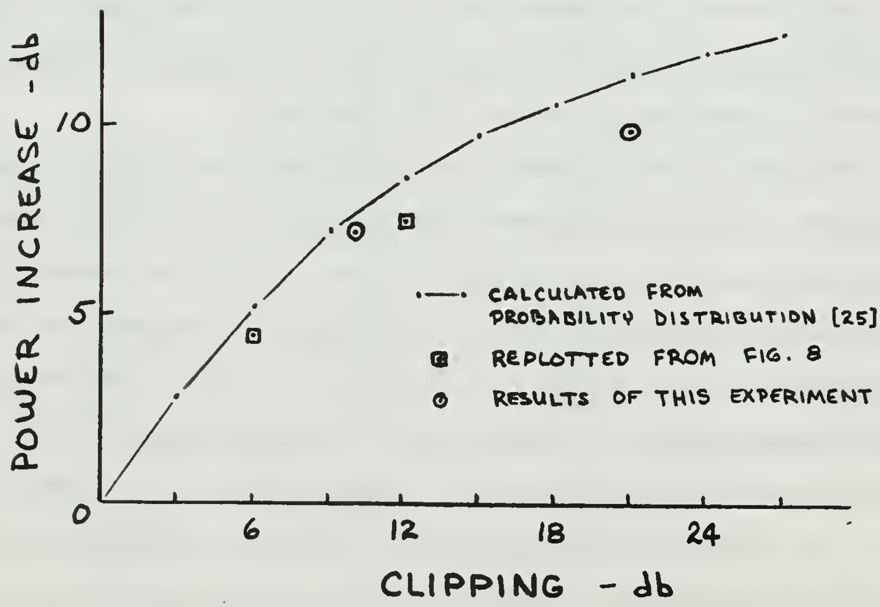


Fig. 20 Total power increase with clipping

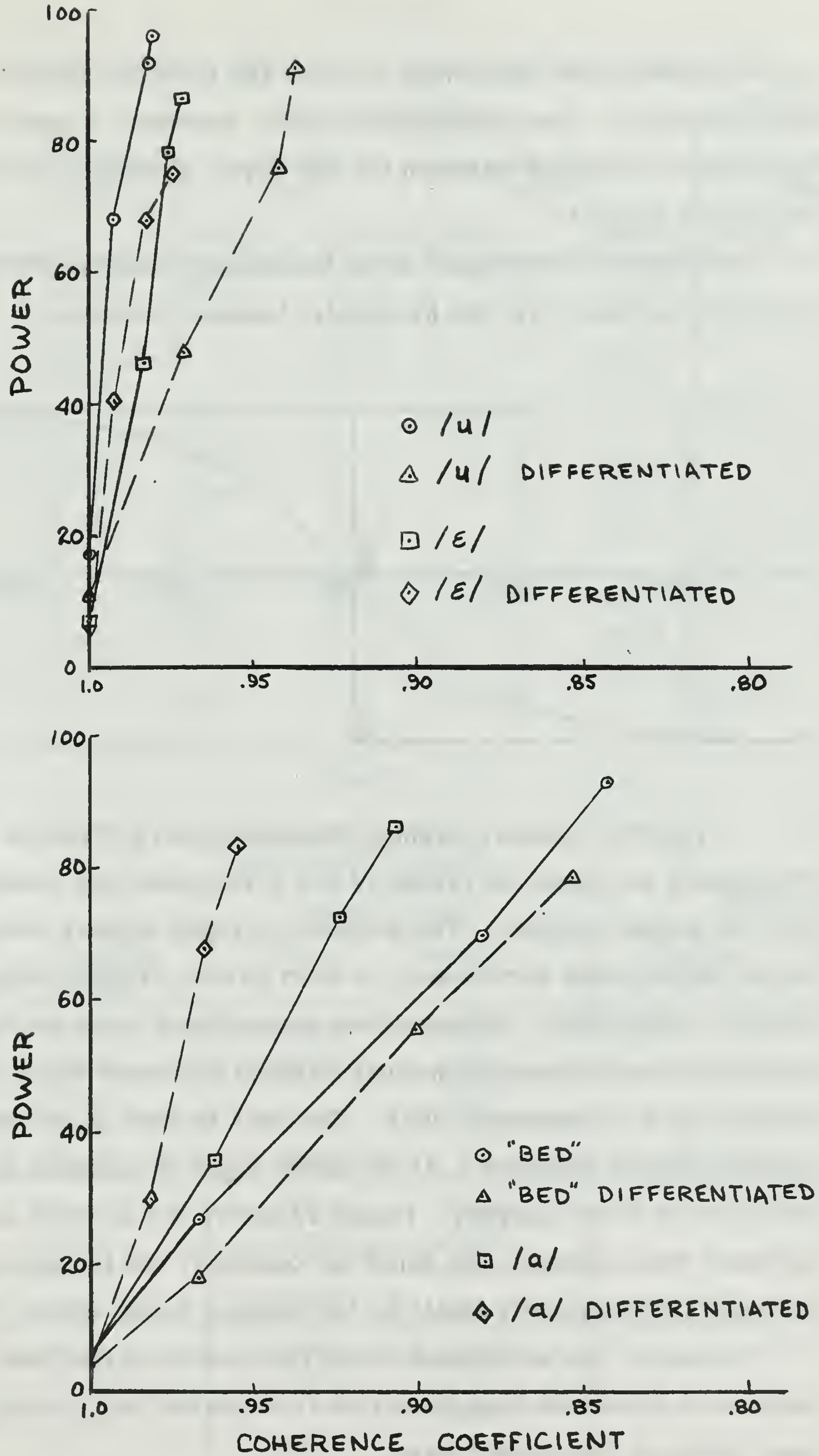


Fig. 21 In-band power vs. coherence coefficient

have rounded rather than sharp corners and therefore should contain less distortion. The intelligibility tests, however, showed no significant difference between the two types of clipper when additive noise was present.

Calculations were made using the gradual clipping characteristic shown in Figure 22: the hyperbolic tangent function.

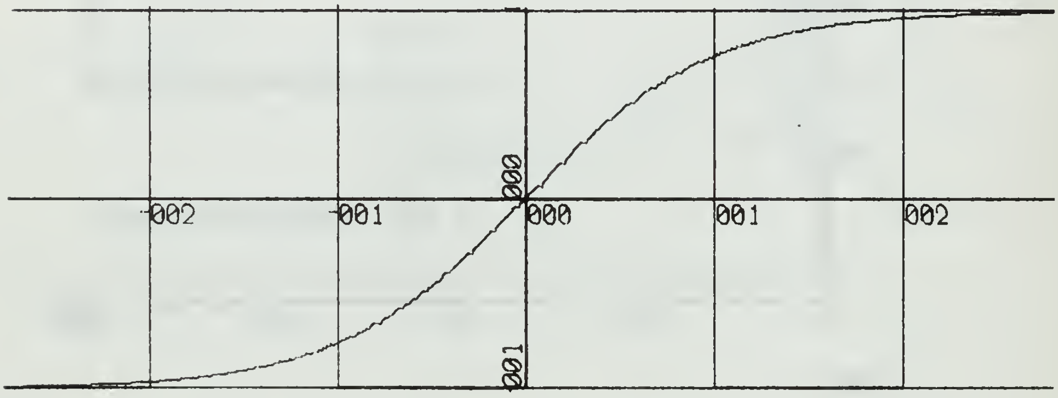


Fig. 22 Gradual clipping characteristic: y = tanh x

The results are listed in Tables IV and V alongside the corresponding data for abrupt clipping. The gradually clipped signals have less power (higher peak factor) and, in most cases, slightly larger coherence coefficients. Where direct comparisons could be made, the coherence coefficients for gradual clipping exceeded those for abrupt clipping by an average of .0058. The peak factors of the gradually clipped signals averaged 1.15 db higher at 10 db clipping and 0.77 db higher at 20 db clipping. Figure 23 shows the in-band power increase with clipping. As could be expected, the increase in power becomes insignificantly small as for clipping levels above 30 db.

Evidently, any advantage which the gradual clipper may have because it generates less distortion is cancelled out by the higher peak factor of the clipped wave.

TABLE IV

Comparison of Power for Gradual and Abrupt Clipping

Sound model	Clipping level	Audio Clipping				SSB Clipping	
		In-band power		Total power		In-band power	
		Abrupt	Gradual	Abrupt	Gradual	Abrupt	Gradual
"bed"	10 db	26.6	21.4	27.3	21.9		
	20 db	69.3	56.3	71.5	58.1		55.5
	30 db		84.1		87.2		
	40 db		92.5		99.7		74.71
	inf	92.5		100.0		75.0	
/a/	10 db	35.6	26.5	35.9	26.8		
	20 db	72.3	62.3	75.8	64.4		62.8
	30 db		79.5		86.1		
	inf	85.9		100.0		77.4	
/a/ differ-entiated	10 db	29.7	22.5	30.3	23.0		
	20 db	67.8	56.9	73.5	60.4		57.4
	30 db		76.8		86.2		
	inf	83.1		100.0		75.6	

TABLE V

Comparison of Coherence Coefficients for Gradual and Abrupt Clipping

Sound model	Clipping level	Audio Clipping		SSB Clipping	
		Abrupt	Gradual	Abrupt	Gradual
"bed"	10 db	.9688	.9679		
	20 db	.8808	.8964		.9050
	30 db		.8525		
	40 db		.8430		.8726
	inf	.8430		.8723	
/a/	10 db	.9623	.9691		
	20 db	.9238	.9323		.9452
	30 db		.9067		
	inf	.9072		.9320	
/a/ differ-entiated	10 db	.9816	.9841		
	20 db	.9647	.9670		.9677
	30 db		.9584		
	inf	.9551		.9604	

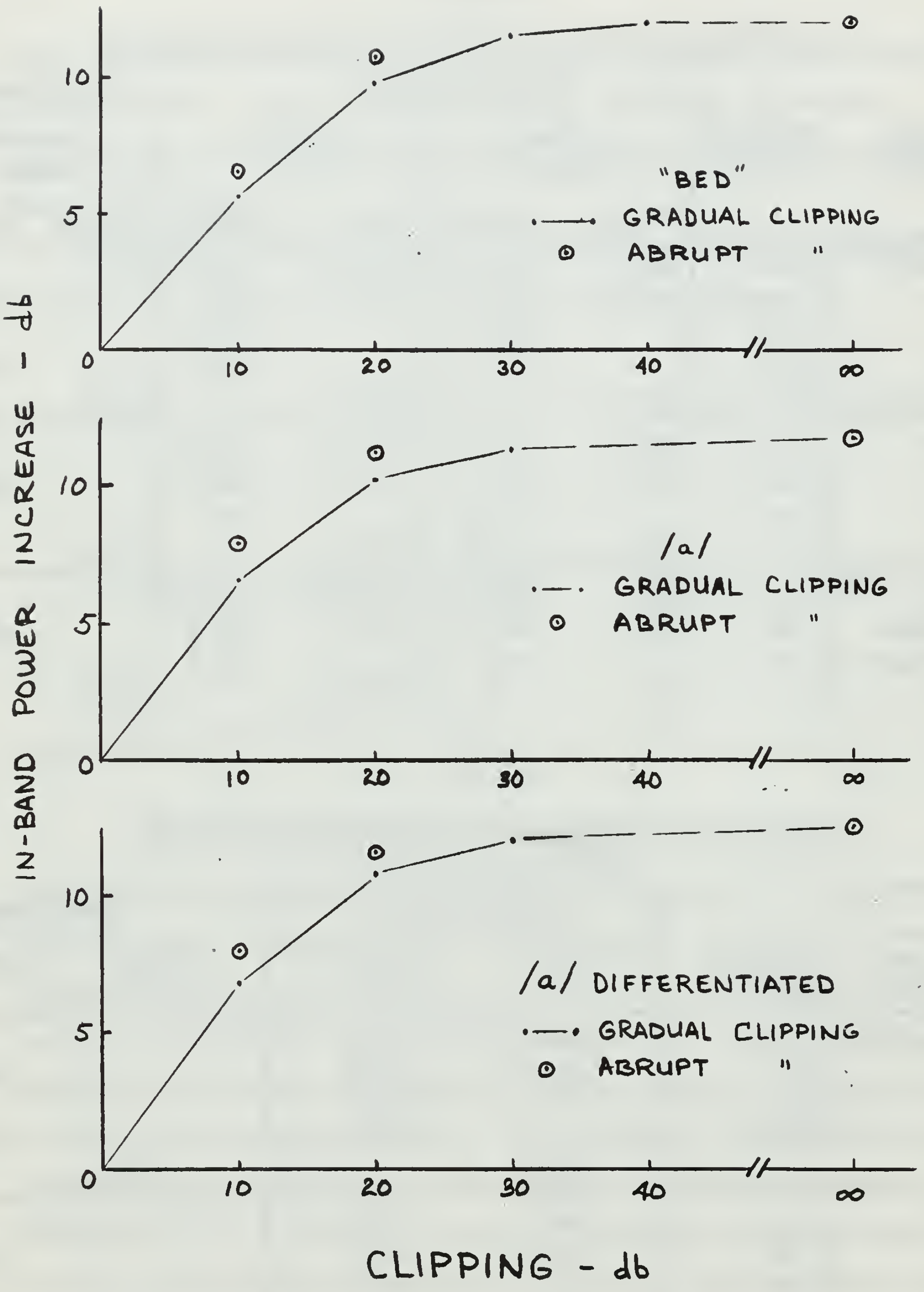


Fig. 23 In-band power increase for gradual and abrupt audio clipping.

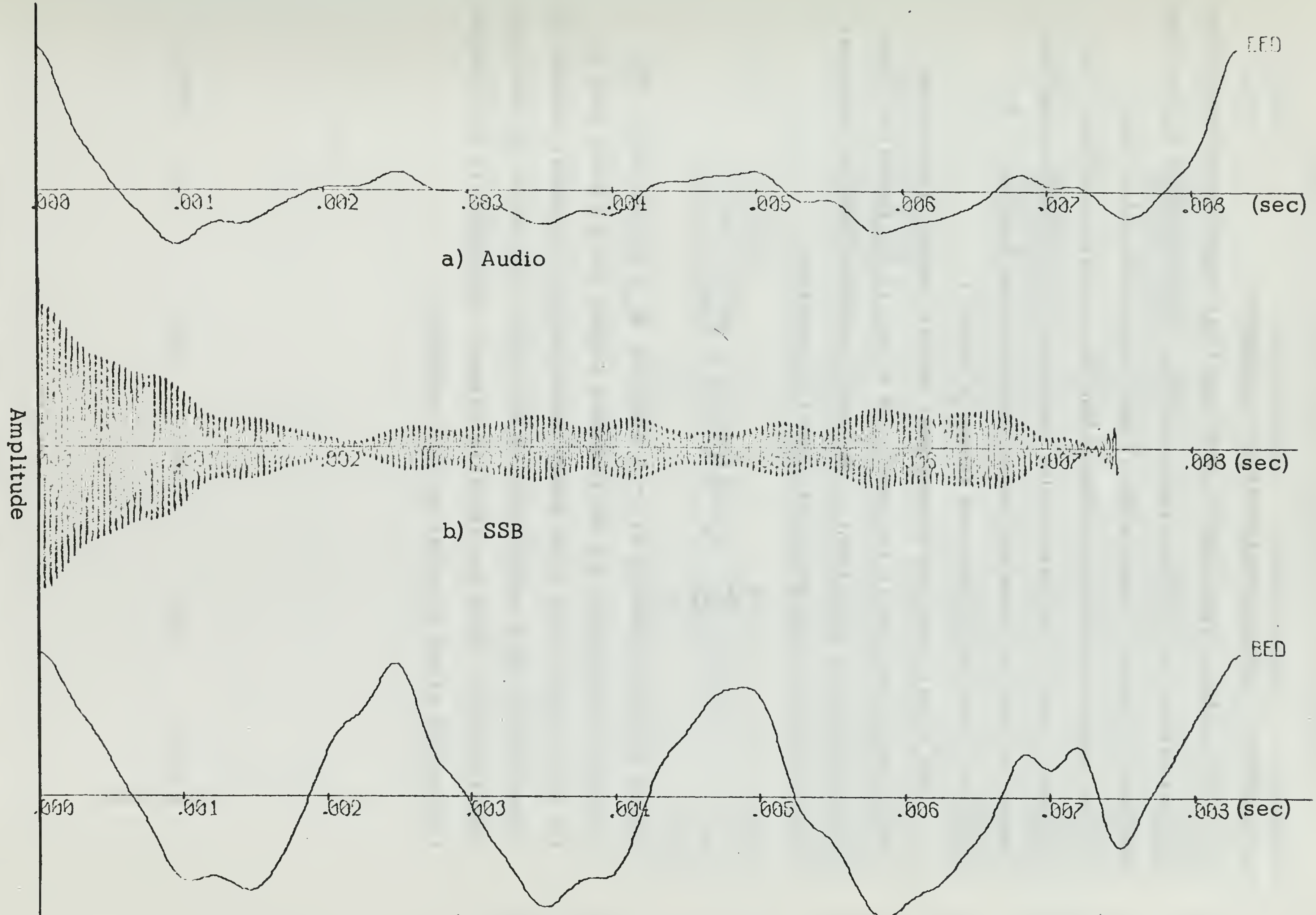
8. Repeaking

It is well known that when a clipped speech signal is filtered to reduce its bandwidth the peak factor increases. One explanation for the increase is obviously that the out-of-band power is removed by filtering leaving, for the same peak value, a lower rms value. Another explanation that has been advanced is that the phase characteristic of the filter causes the components to add in such a way that the signal repeaks. In Section 7 repeaking due to elimination of the out-of-band power was taken into account by computing the power remaining in the band of the original signal.

To investigate repeaking further, the spectrum of the clipped signal was transformed back to the time domain using only the components in the band of the original signal. The process is equivalent to passing the clipped signal through an ideal filter having a rectangular amplitude characteristic and a linear phase characteristic. For the case of infinite SSB clipping, calculations were made using the sound models "bed" and /ε/. The peak values of the time waveforms increased (from 10) to 16.85 for "bed" and 16.75 for /ε/. The resulting peak factor was 5.72 db in both cases. For infinite clipping at audio followed by ideal filtering, the average peak factor for the four sound models was 3.17 db.

These results might seem at first glance to indicate a significant advantage for audio clipping: a smaller resultant peak factor. Upon closer examination, however, the large difference in peak factor disappears. The narrowband signal which has the smallest possible peak factor is a simple sinusoid of constant amplitude. If this sinusoid were to be altered so as to reduce its peak factor, for example by making it a rectangular wave, the altered signal would contain harmonics at multiples of the sinusoid frequency and thus would no longer be a narrowband signal. Therefore the smallest peak factor that an SSB signal can have is 3 db. The peak factor computed for the clipped SSB signals is only 2.72 db above this minimum.

It has been assumed in various studies [21, 24] that the peak factor of a signal changes significantly with frequency translation from audio to SSB. Three calculations were made in order to check this assumption. When the unclipped model sounds "bed", /a/, and /ɛ/ were translated to narrowband, the peak factors changed only +0.27 db, +0.65 db, and -0.03 db respectively. This is a rather surprising result considering that the frequency components which are harmonically related at audio are of nearly the same frequency at narrowband. When two of the clipped and filtered SSB signals, "bed", and /ɛ/, were translated back to audio, the peak factor changed by +0.01 db and -0.48 db respectively. Shown in Figure 24 are the time waveforms for "bed" at audio and SSB.



a) Audio

b) SSB

c) Retranslated to audio after infinite clipping and filtering at SSB

Fig. 24 Time waveforms for "bed" at audio and SSB

9. Effects of phase characteristics on clipping

Calculations were made to determine what effect the original phase characteristic has on the clipping process. The four model sounds were infinitely clipped at audio with: 1) their components all in phase, 2) the phase characteristic inferred from the spectrum amplitude distribution, and 3) the phase angles determined by a random number generator so that they were uniformly distributed from 0 to 2π. The results of the calculations are shown in Table VI.

The signals have the largest peak factors before clipping when their components are all in phase. (PF = 20 - 10 $\log_{10}$(Power) db since the signals were normalized so that their maximum absolute value was 10.) This result was expected since the expression

$$f(t) = \sum_{n=1}^{N} C_n \cos(2\pi n f_o t + \phi_n)$$

has an absolute maximum at t = 0 when the ϕ_n's are all zero. The coherence coefficients for the in-phase signals are all lower than for the same signals with other phase characteristics, probably because the large positive peak causes much of the signal to be below the zero axis. (See Figure 25) When the signal is clipped, information is lost because this portion of the wave has no zero crossings.

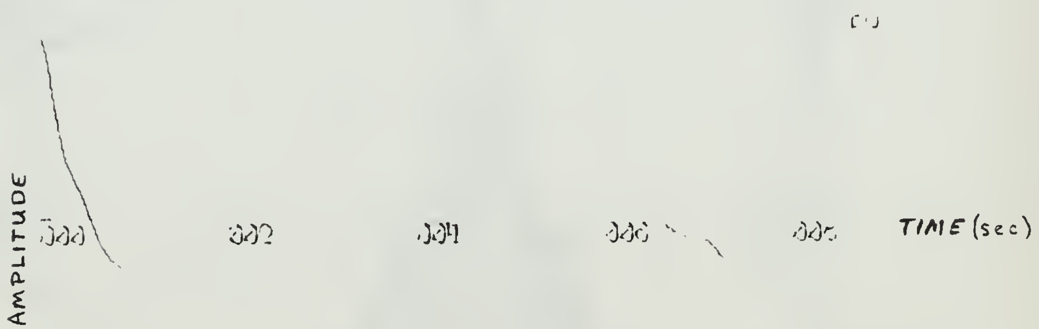


Fig. 25 Time waveform for "bed" with all components in phase

TABLE VI

Effects of Original Phase Characteristic on Clipping

Sound model		Power before clipping	In-band power after clipping	Coherence coefficient
"bed"	In phase	4.7	45.4	.7461
	Phase inferred	5.9	92.5	.8430
	Random phase	15.2	89.4	.9244
		12.4	83.9	.8698
		22.1	93.7	.9042
/ɛ/	In phase	3.4	88.4	.9193
	Phase inferred	7.7	86.7	.9725
	Random phase	11.8	83.7	.9844
		19.4	83.0	.9857
		18.5	85.3	.9798
/a/	In phase	3.2	84.5	.7665
	Phase inferred	5.8	85.9	.9071
	Random phase	18.1	87.3	.9003
		21.2	86.3	.9433
		19.3	88.0	.8998
/u/	In phase	9.8	93.6	.9375
	Phase inferred	17.3	95.9	.9805
	Random phase	18.7	95.2	.9879
		22.8	95.6	.9840
		16.4	95.2	.9758

With the phase characteristic inferred from the spectral amplitude distribution, the peak factors are smaller and the coherence coefficients larger. With random phases the peak factors decrease even further. Since the ear is quite insensitive to phase variations [6], this result suggests that the peak factor of speech could be reduced, without imparing intelligibility, by passing it through some non-linear phase network which would not distort the spectrum appreciably. For any signal expressed in the form of Eq. 9.1, there is some phase characteristic which minimizes its maximum absolute value. The questions of what that phase characteristic is, and whether or not an approximation to it can be obtained with a realizable network are subjects for further research in this area. Notice also that the coherence coefficients are generally larger for the random phase signals. This indicates that the intelligibility of clipped speech might be enhanced by some phase variation before clipping.

10. Conclusion

The analysis of the effects of peak clipping on speech-like signals has produced three important findings. First, the coherence coefficients are large, indicating a high degree of similarity between the power spectra before and after clipping. Thus it is not so surprising that a speech signal subjected to the severe amplitude distortion of infinite clipping can be understood. Second, the peak factor did not change appreciably with frequency translation in two specific instances: the translation of an unclipped audio signal to narrowband and the translation of an infinitely clipped, ideally filtered signal back to audio. Third, it may be possible to reduce the peak factor of unprocessed speech significantly by alteration of its phase characteristic. An appropriate variation in phase before clipping may also enhance the intelligibility of clipped speech by increasing the coherence between the spectra before and after clipping.

Although experimental data on actual vowel sounds by themselves was not available for direct comparison, the calculated results are in general agreement with published figures. The real measure of effectiveness of any speech processing system is in terms of intelligibility of the processed speech in noise, but the method of analysis used in this study can give indications which point toward improved speech processing schemes.

BIBLIOGRAPHY

1. Bogert, B.P. "On the Band Width of Vowel Formants," Journal of the Acoustical Society of America, v. 25, (July 1953), 791-792.

2. Chang, S., Phil, G.E., and Essigmann, M.W. "Representation of Speech and Sounds and Some of Their Statistical Properties," Proceedings of the Institute of Radio Engineers, v. 39, (February 1951), 147-153.

3. Davenport, W.B. "An Experimental Study of Speech Wave Probability Distributions," Journal of the Acoustical Society of America, v. 24, (July 1952), 390-399.

4. Dukes, J.M.C. "The Effect of Severe Amplitude Limitation on Certain Types of Random Signal; A Clue to the Intelligibility of 'Infinitely' Clipped Speech," Proceedings of the Institute of Electrical Engineers (London), v. 102-103, Pt. C, 1955-56, 88-97.

5. Fant, G. Acoustic Theory of Speech Production, 'S Gravenhage: Mouton & Co., 1960.

6. Fletcher, H. Speech and Hearing in Communication. New York: D. Van Nostrand Company, 1953.

7. French, N., and Steinberg, J. "Factors Covering the Intelligibility of Speech Sounds," Journal of the Acoustical Society of America, v. 19, (January 1947), 90-119.

8. Huddy, N.W., Jr. "An Investigation of Methods of Improving the Intelligibility of Audio Frequency Speech in Noise." Master's thesis, Naval Postgraduate School, Monterey, California, 1966.

9. Licklider, J. "Amplitude Distortion and Speech Intelligibility", Journal of the Acoustical Society of America, v. 18, 1946, 429-434.

10. Licklider, J., and Pollack, I. "Effects of Differentiation, Integration, and Infinite Peak Clipping on the Intelligibility of Speech," Journal of the Acoustical Society of America, v. 20, (January 1948), 42-51.

11. Middleton, D., and Van Vleck, J. "The Spectrum of Clipped Noise," Proceedings of the Institute of Electrical and Electronics Engineers, v. 54, (January 1966), 2-19.

12. Miller, R.L. "Nature of the Vocal Card Wave," Journal of the Acoustical Society of America, v. 31, (June 1959), 667-677.

13. Office of Scientific Research and Development, The Effects of Amplitude Distortion upon the Intelligibility of Speech, OSRD Report No. 4217, Harvard University, 1944.

14. Pappenfuss, E., Bruene, W., and Schenike, E. Single Sideband Principles and Circuits. New York: McGraw-Hill Book Company, 1964.

15. Pollack, I. "On the Effect of Frequency and Amplitude Distortion on the Intelligibility of Speech in Noise," Journal of the Acoustical Society of America, v. 24, (September 1952), 538-540.

16. Pollack, I., and Pickett, J. "Intelligibility of Peak-Clipped Speech at High Noise Levels," Journal of the Acoustical Society of America, v. 31, (January 1959), 14-16.

17. Rosen, G. "Dynamic Analog Speech Synthesizer," Journal of the Acoustical Society of America, v. 30, (March 1958), 201-209.

18. Rostovtsev, Y.G. "The Possibility of Using Maximum Amplitude Limitation of Speech Signals in Systems of Communication," Telecommunications, 1958, 643-648.

19. Schroeder, M. "Vocoders: Analysis and Synthesis of Speech," Proceedings of the Institute of Electrical and Electronics Engineers, v. 54, (May 1966), 720-733.

20. Schwarzlander, H. "Intelligibility Evaluation of Voice Communications," Electronics, (May 29, 1959), 88-91.

21. Shyne, N.A. Speech Signal Processing and Applications to Single Side-Band. AFCRL 62-64, Montana State College, 1963.

22. Stevens, K.N., and Fant, C.G.M. "An Electrical Analog of the Vocal Tract," Journal of the Acoustical Society of America, v. 25, (July 1953), 734-742.

23. Stevens, K.N. et al "Speech Analysis and Synthesis Final Report," Massachusetts Institute of Technology. AFCRL 64-300, December 1963.

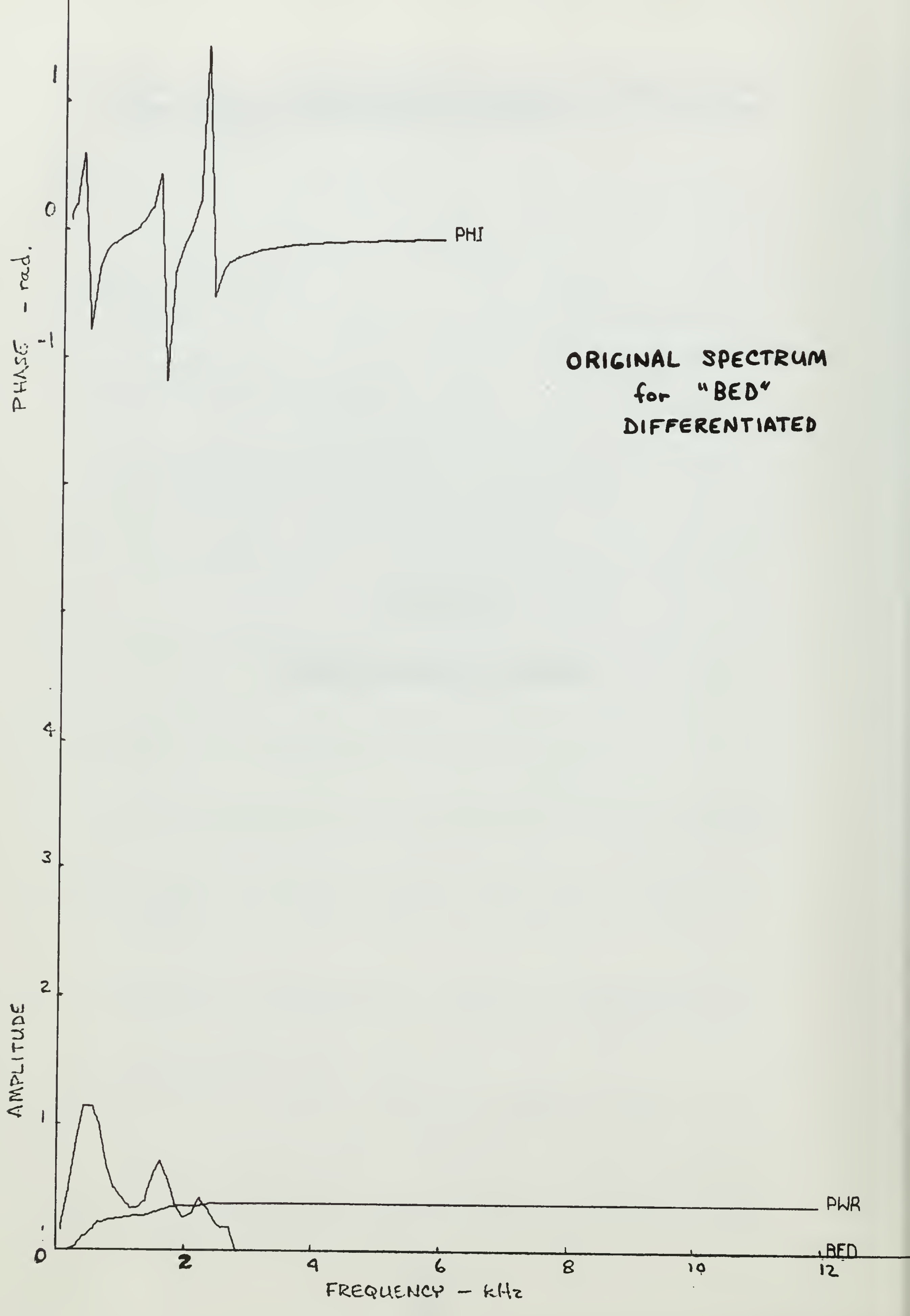
24. U.S. Army Signal Corps. Premodulation and Postmodulation Clipping in Single Sideband Transmission. AEPG-Sig. 960-86, University of Arizona, 1960.

25. Wathen-Dunn, W., and Lipke, P. "On the Power Gained by Clipping Speech in the Audio Band," Journal of the Acoustical Society of America, v. 30, (January 1958), 36-40.

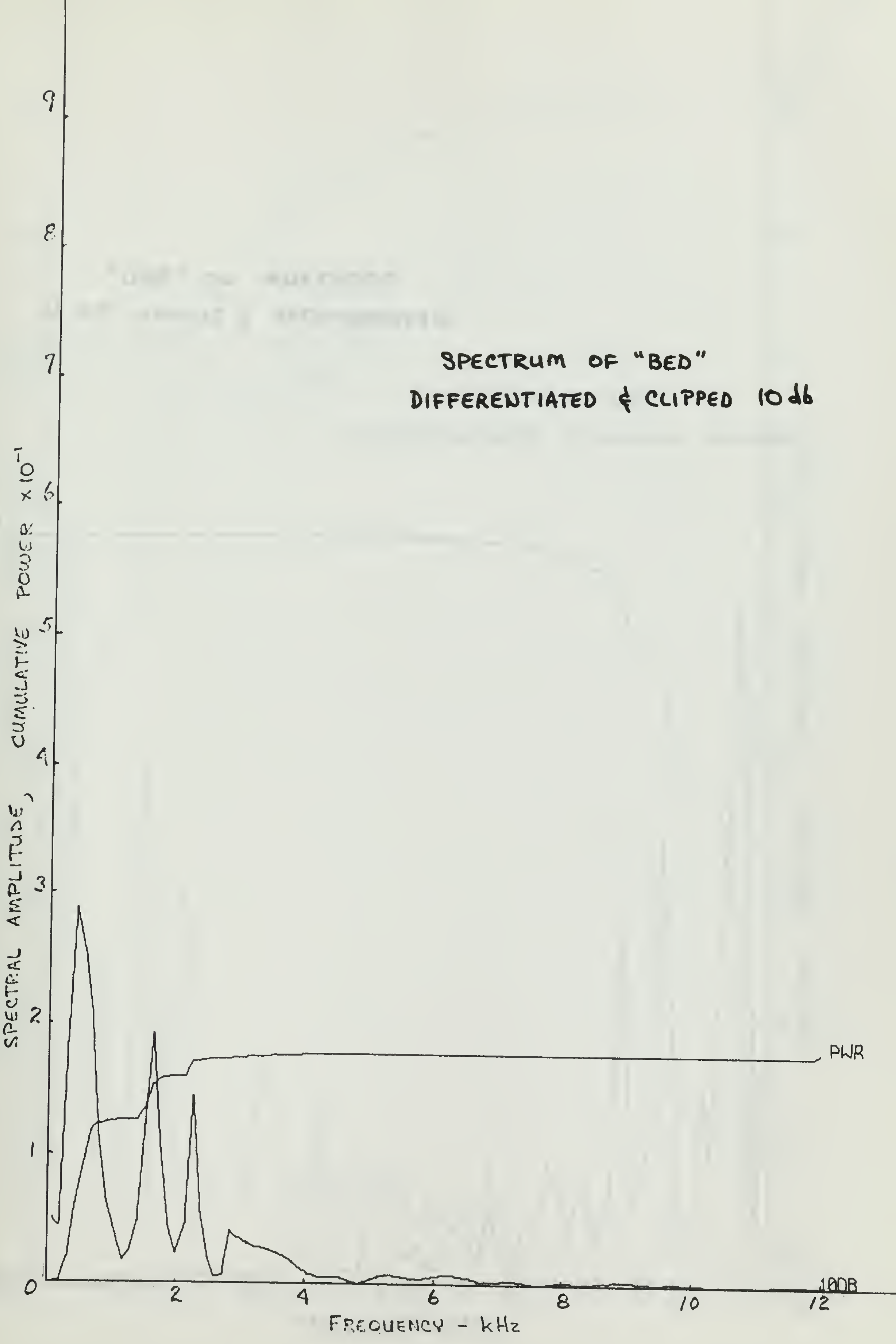
26. Weibel, E.S. "Vowel Synthesis by means of Resonant Circuits," Journal of the Acoustical Society of America, v. 27, (September 1955), 858-865.

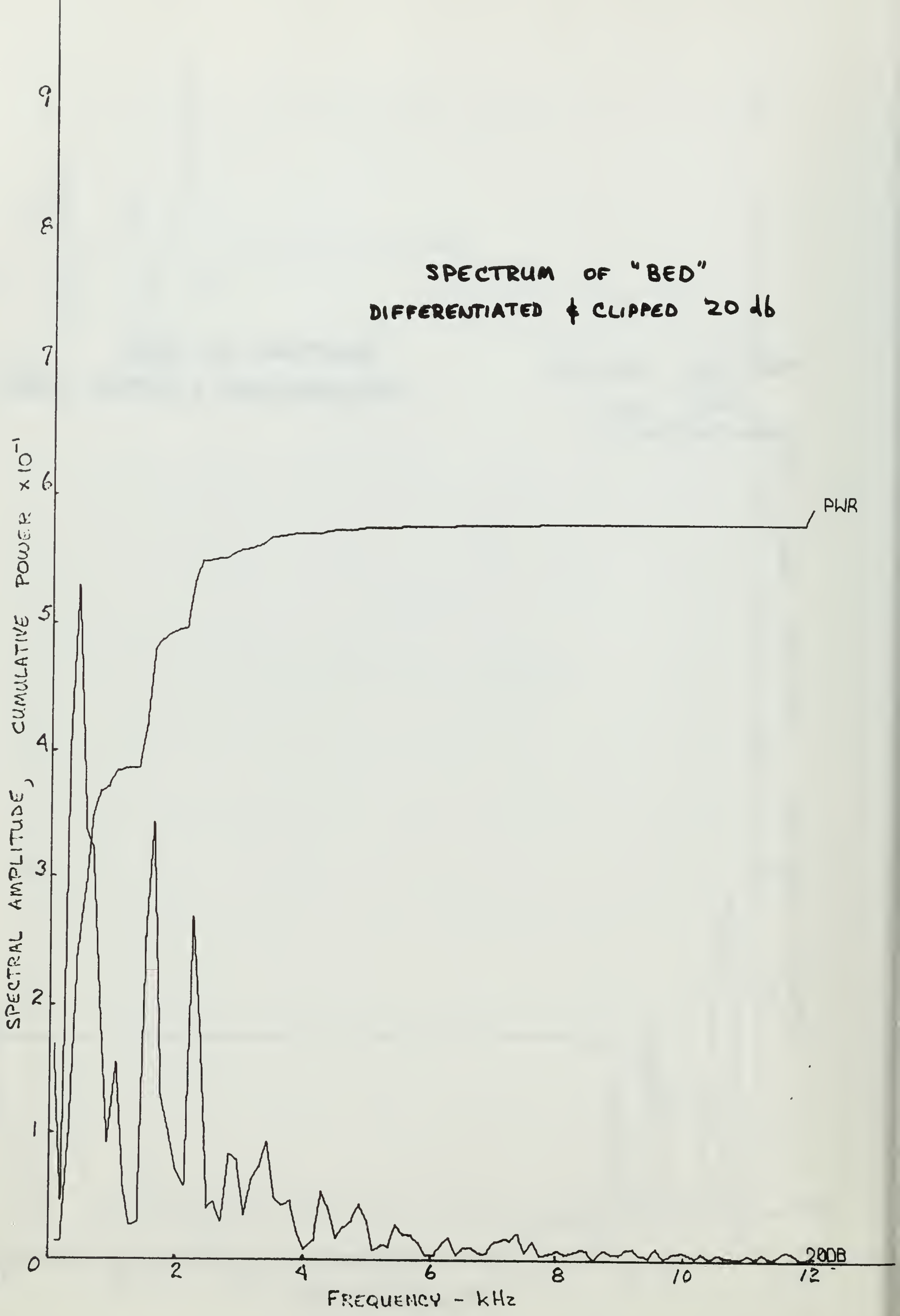
APPENDIX A

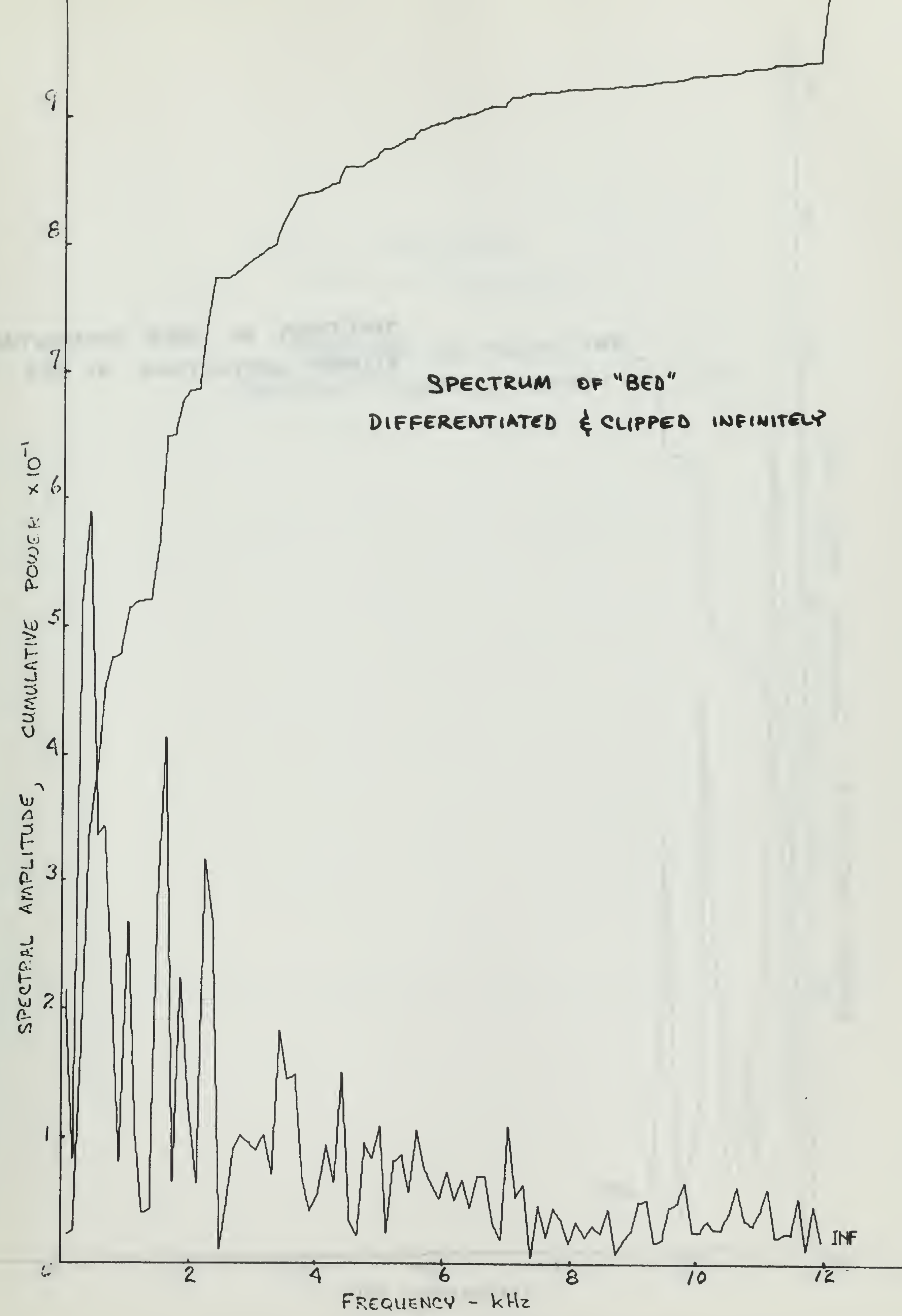
SPECTRA OF CLIPPED SIGNALS

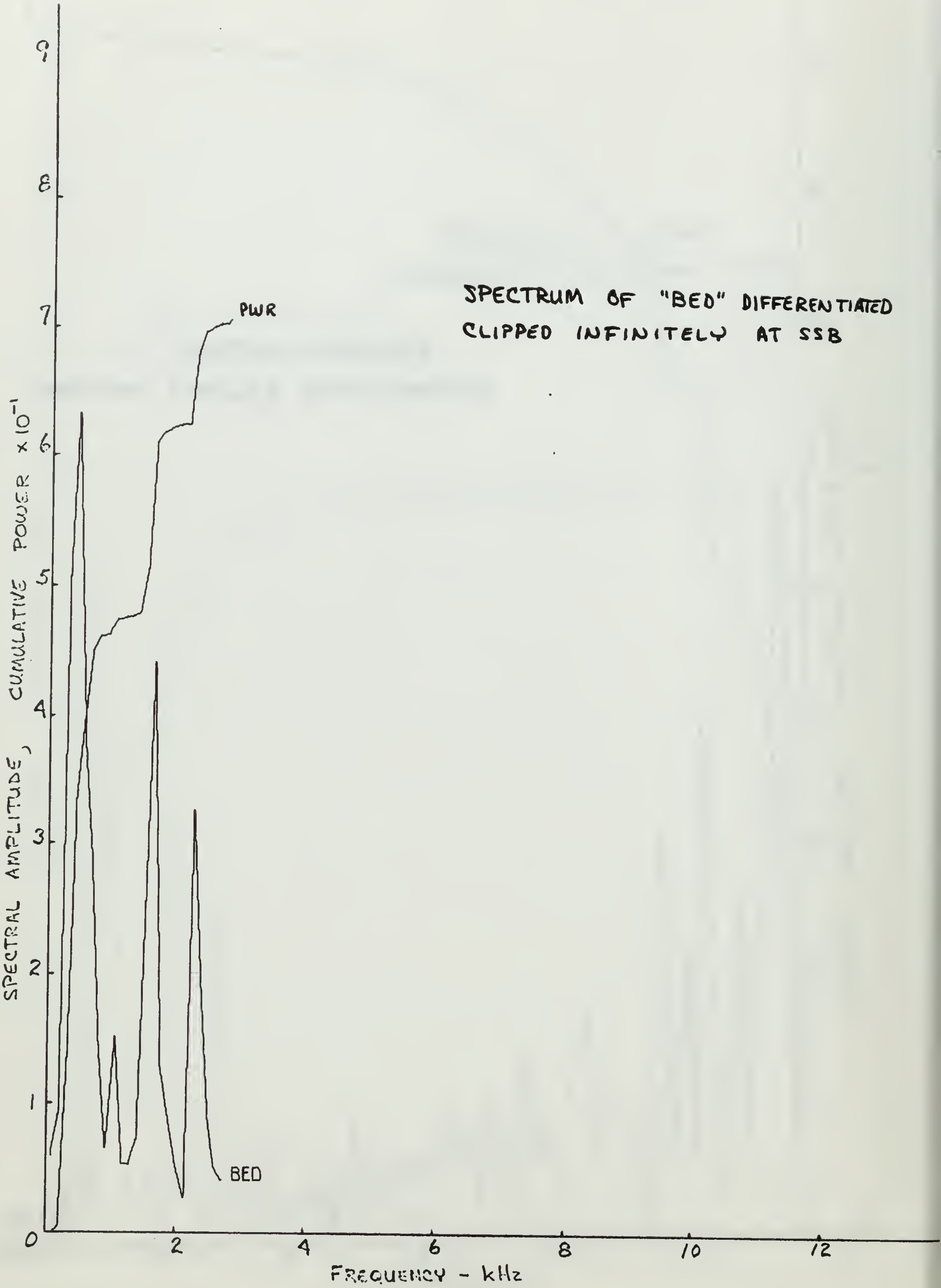


ORIGINAL SPECTRUM
for "BED"
DIFFERENTIATED
PHASE - rad.
1
0
-1
PHI
AMPLITUDE
4
3
2
1
0
PWR
BED
2
4
6
8
10
12
FREQUENCY - kHz

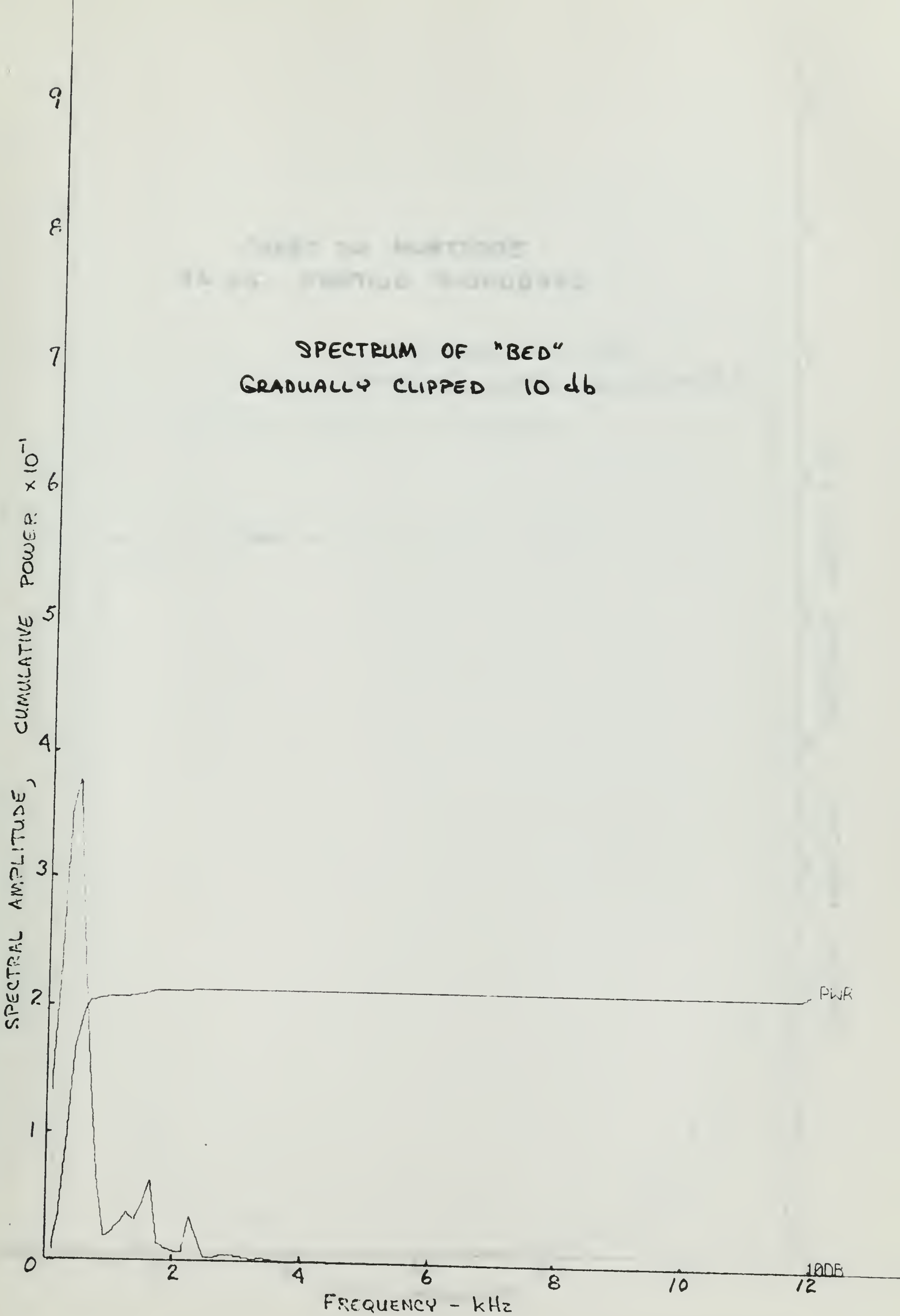

SPECTRUM OF "BED"
DIFFERENTIATED & CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10⁻¹
PWR
10DB
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF "BED"
DIFFERENTIATED & CLIPPED 20 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
PWR
20DB
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
10
12

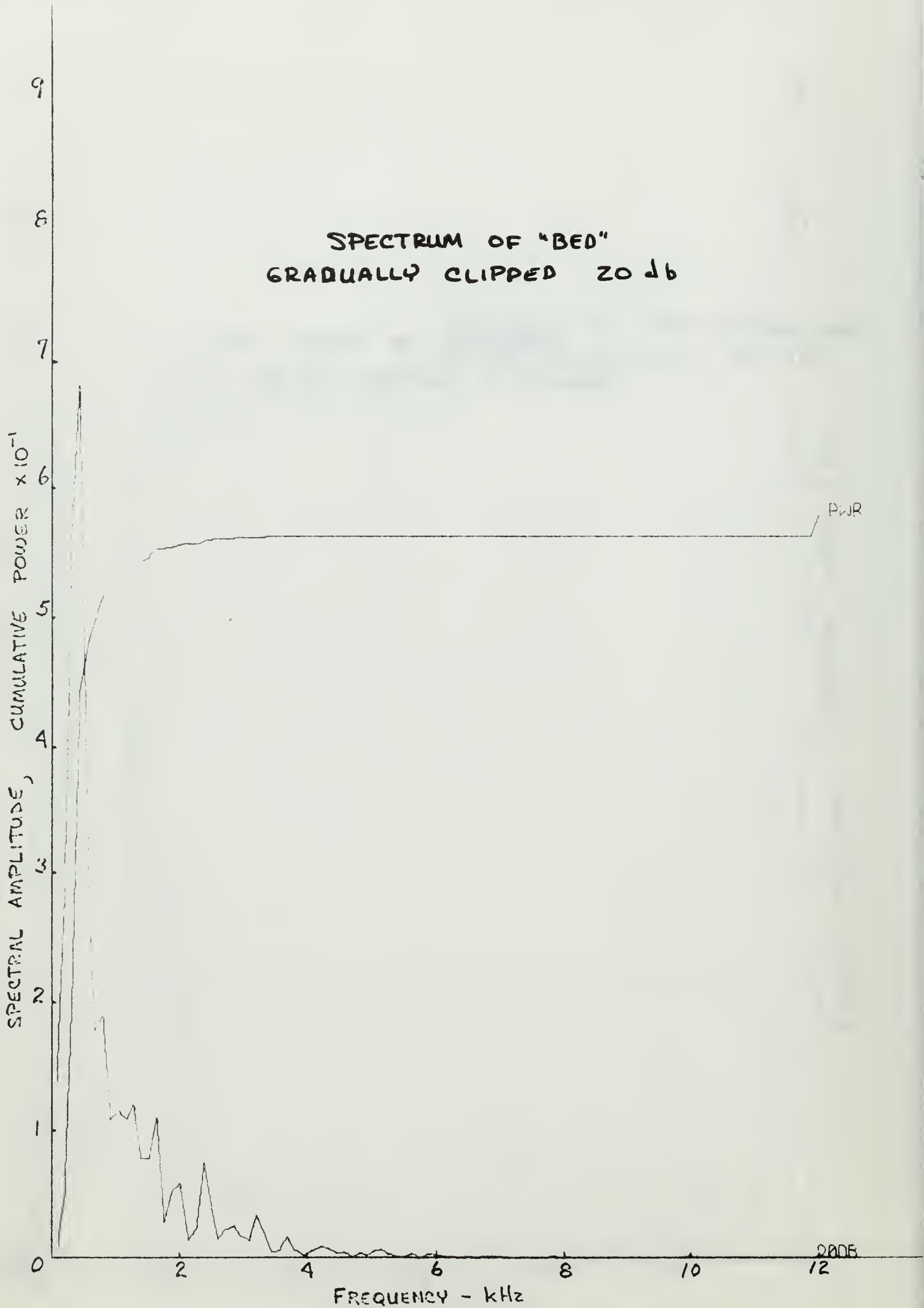

SPECTRUM OF "BED"
DIFFERENTIATED & CLIPPED INFINITELY
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
INF
0
1
2
3
4
5
6
7
8
9
10
12

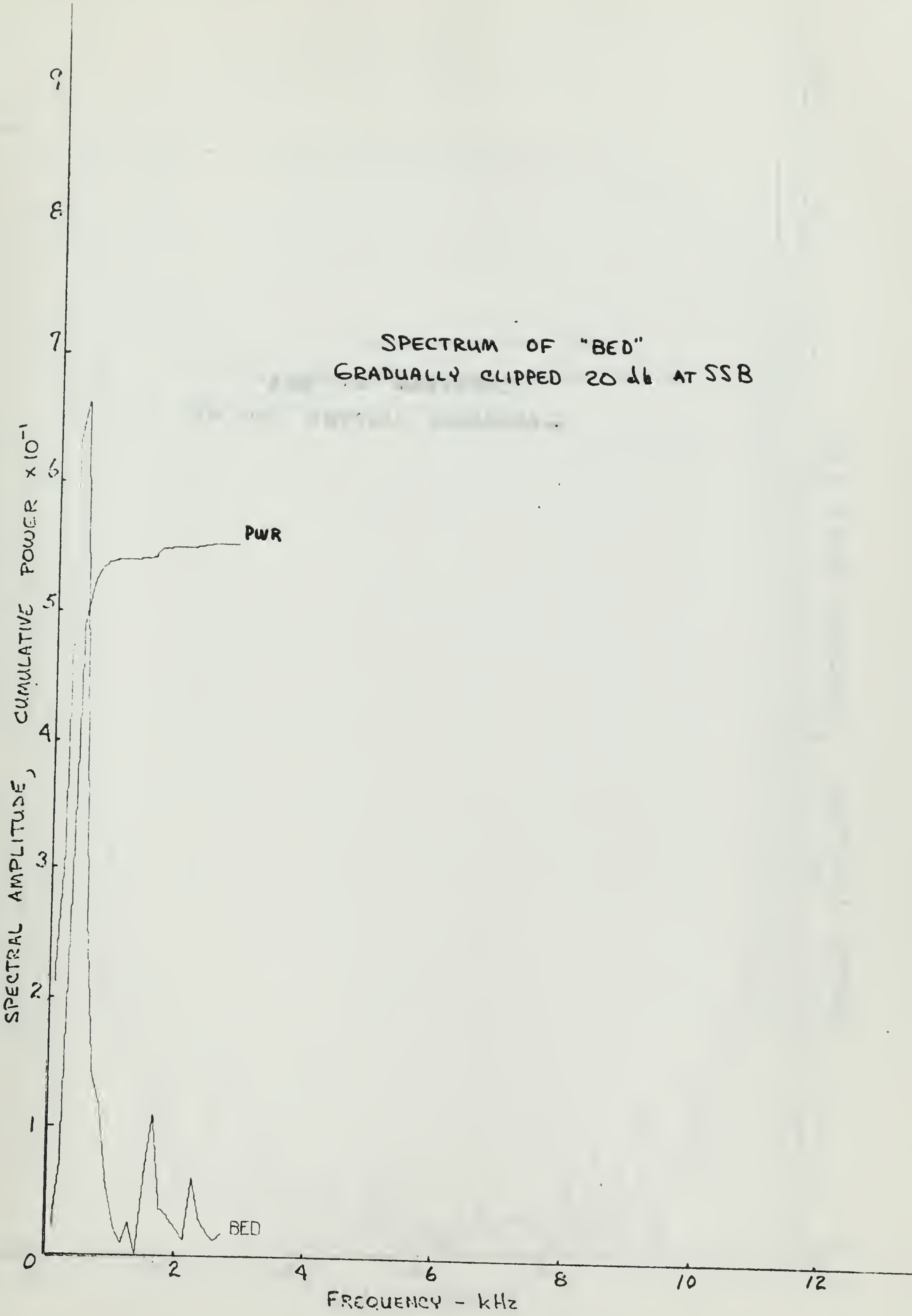


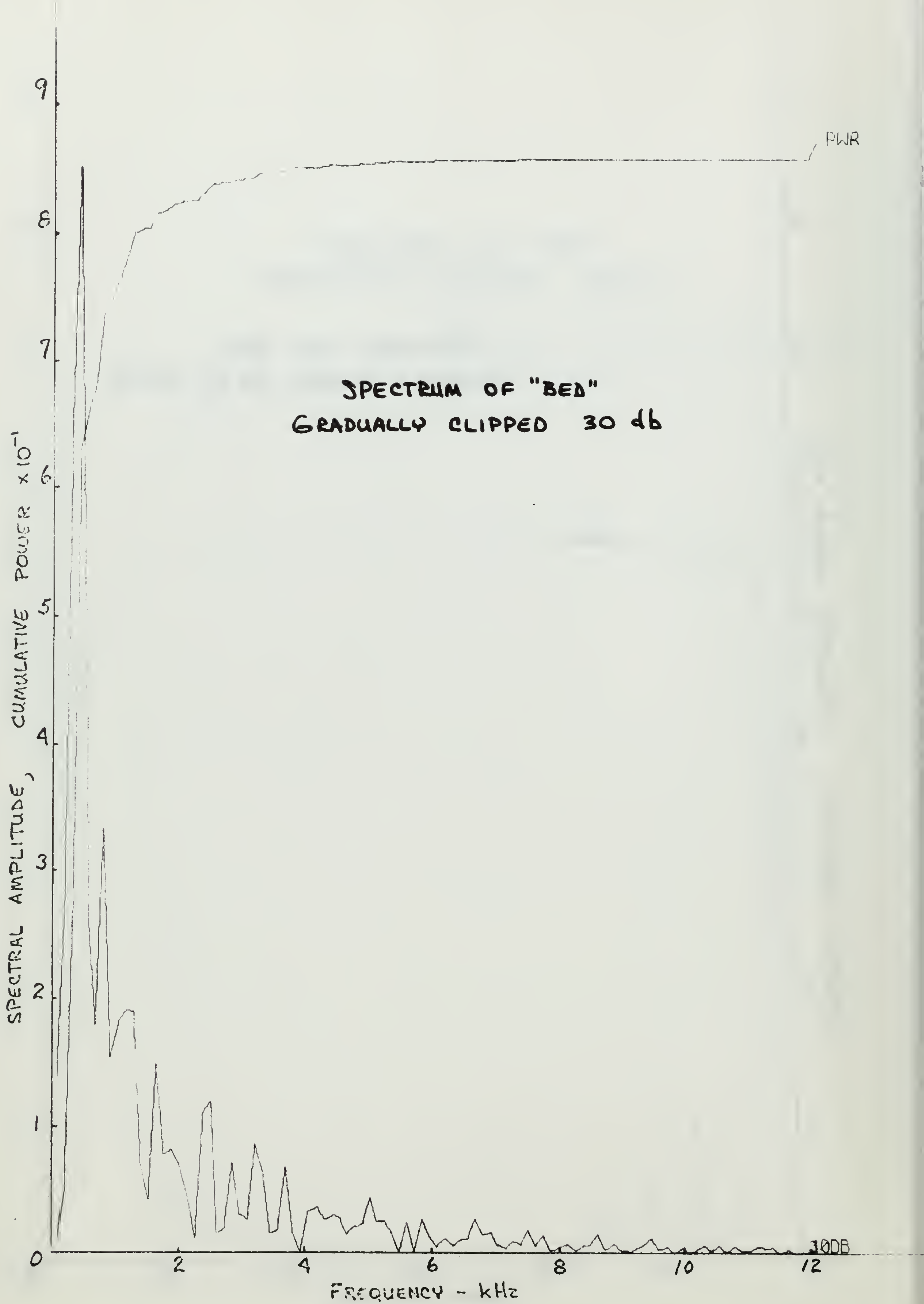
SPECTRUM OF "BED" DIFFERENTIATED
CLIPPED INFINITELY AT SSB
PWR
BED
SPECTRAL AMPLITUDE, CUMULATIVE POWER x 10⁻¹
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12
FREQUENCY - kHz



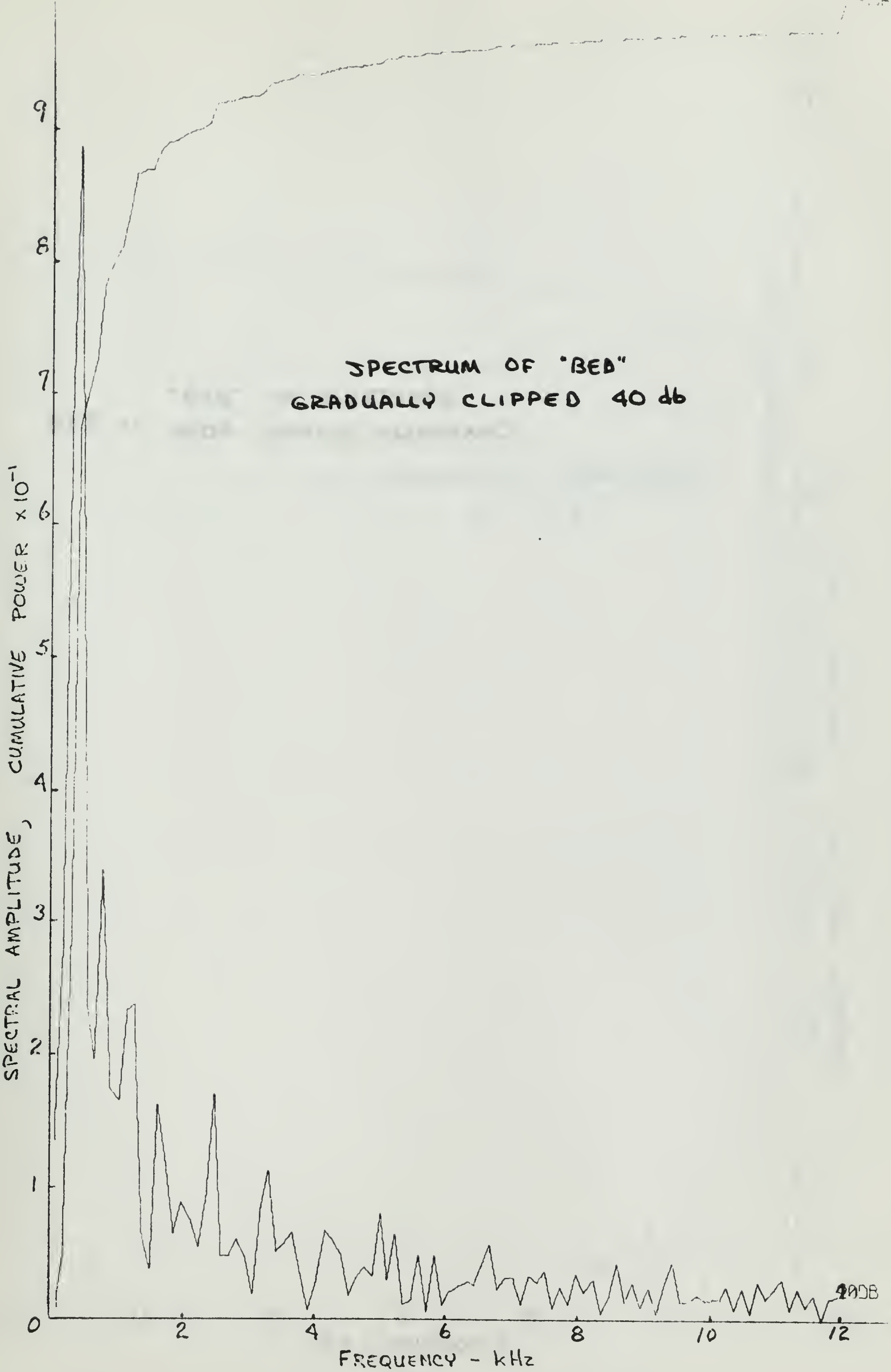
SPECTRUM OF "BED"
GRADUALLY CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
9
8
7
6
5
4
3
2
1
0
2
4
6
8
10
12
FREQUENCY - kHz
PWR
10DB

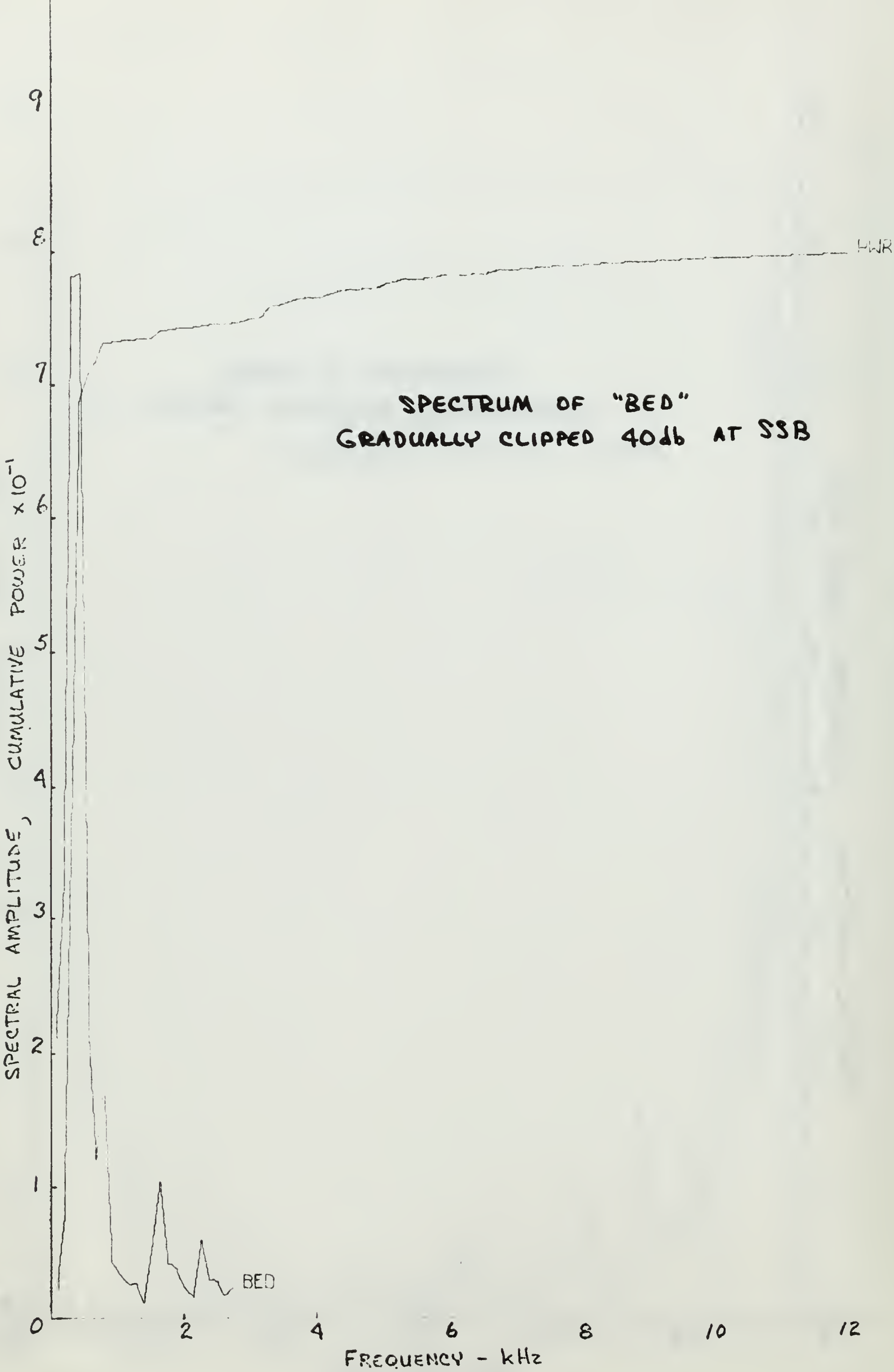

SPECTRUM OF "BED"
GRADUALLY CLIPPED 20 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
PWR
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF "BED"
GRADUALLY CLIPPED 20 db AT SSB
PWR
BED
SPECTRAL AMPLITUDE, CUMULATIVE POWER $\times 10^{-1}$
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

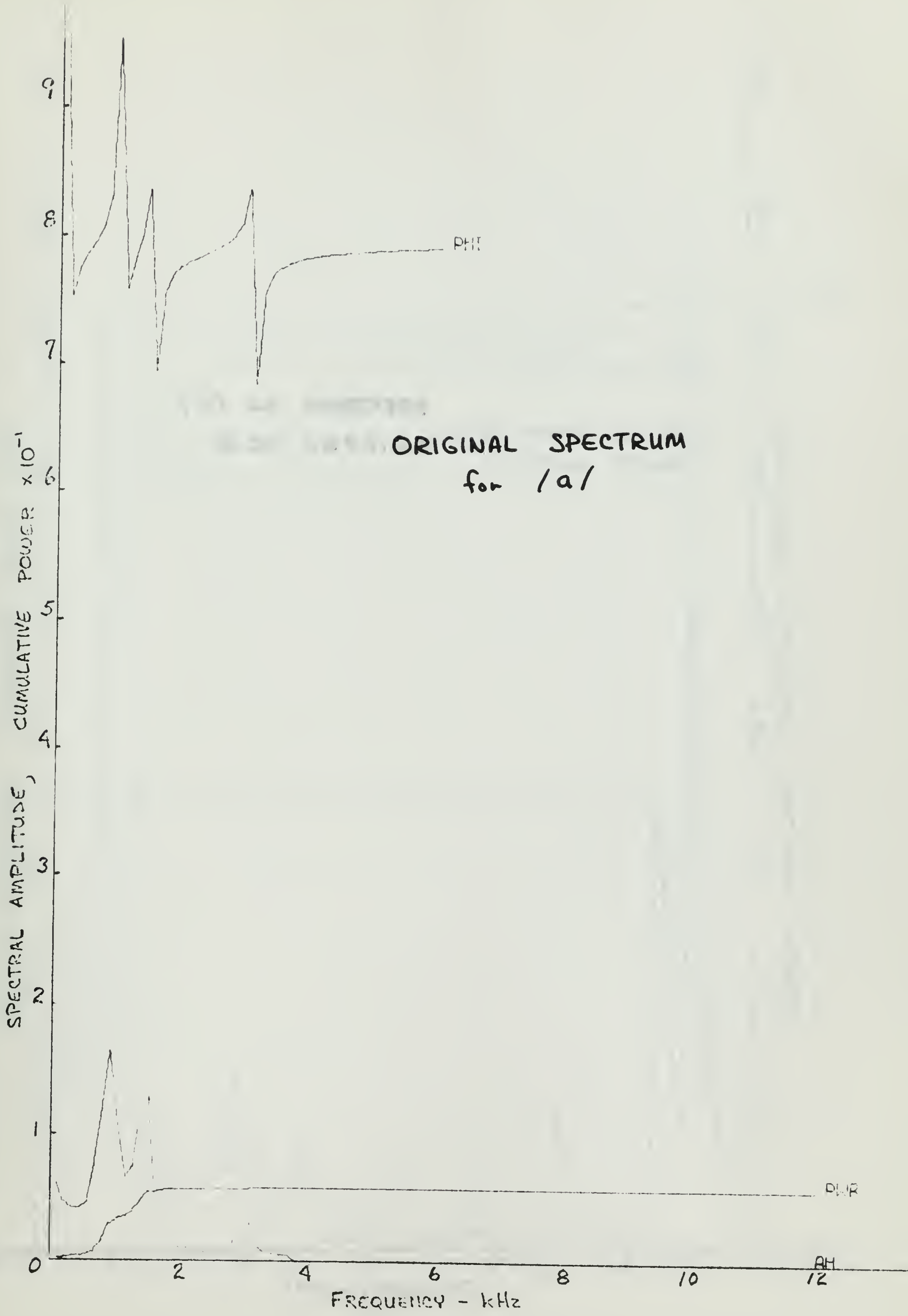


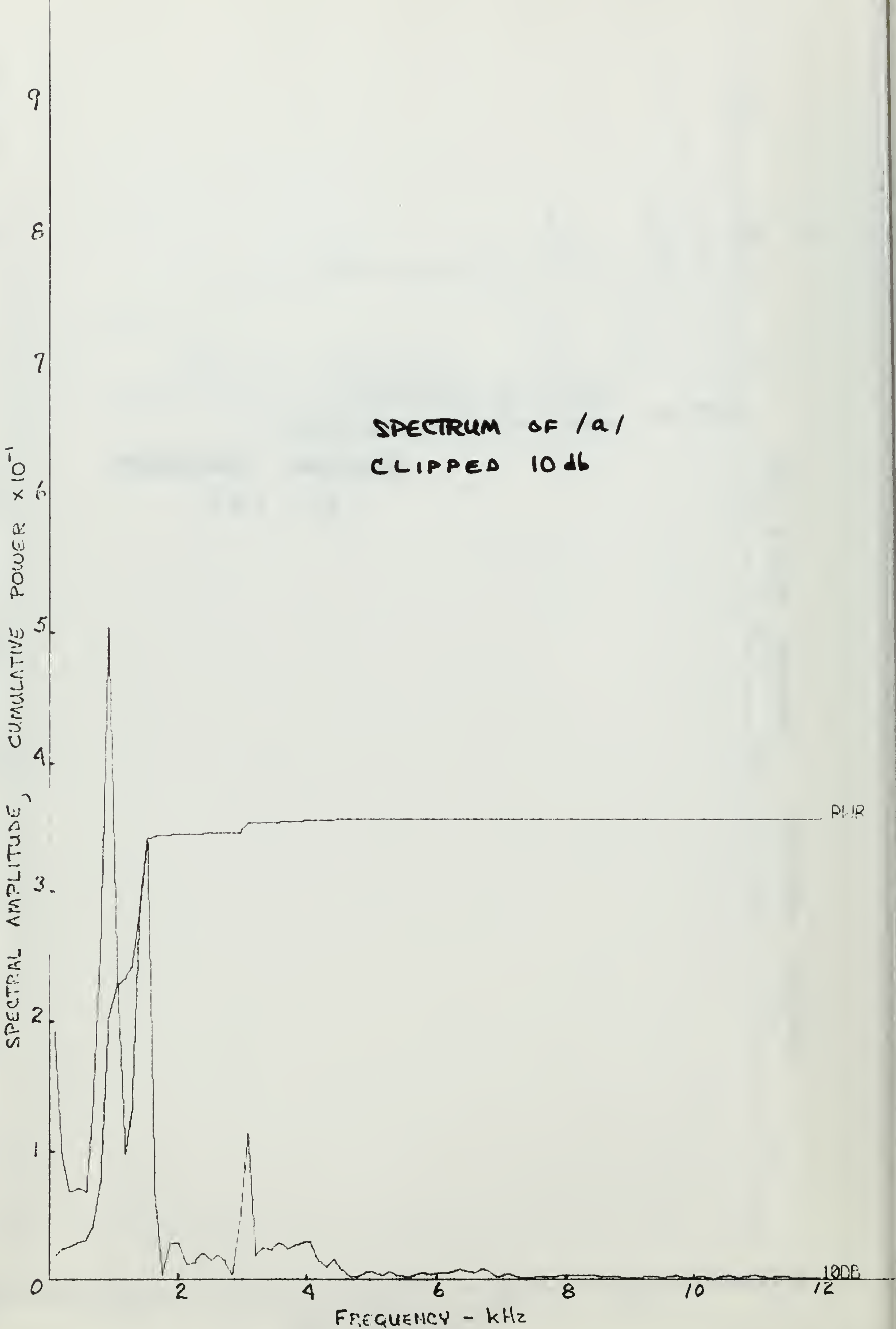
SPECTRUM OF "BED"
GRADUALLY CLIPPED 30 db
PWR
30DB
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
9
8
7
6
5
4
3
2
1
0
2
4
6
8
10
12
FREQUENCY - kHz

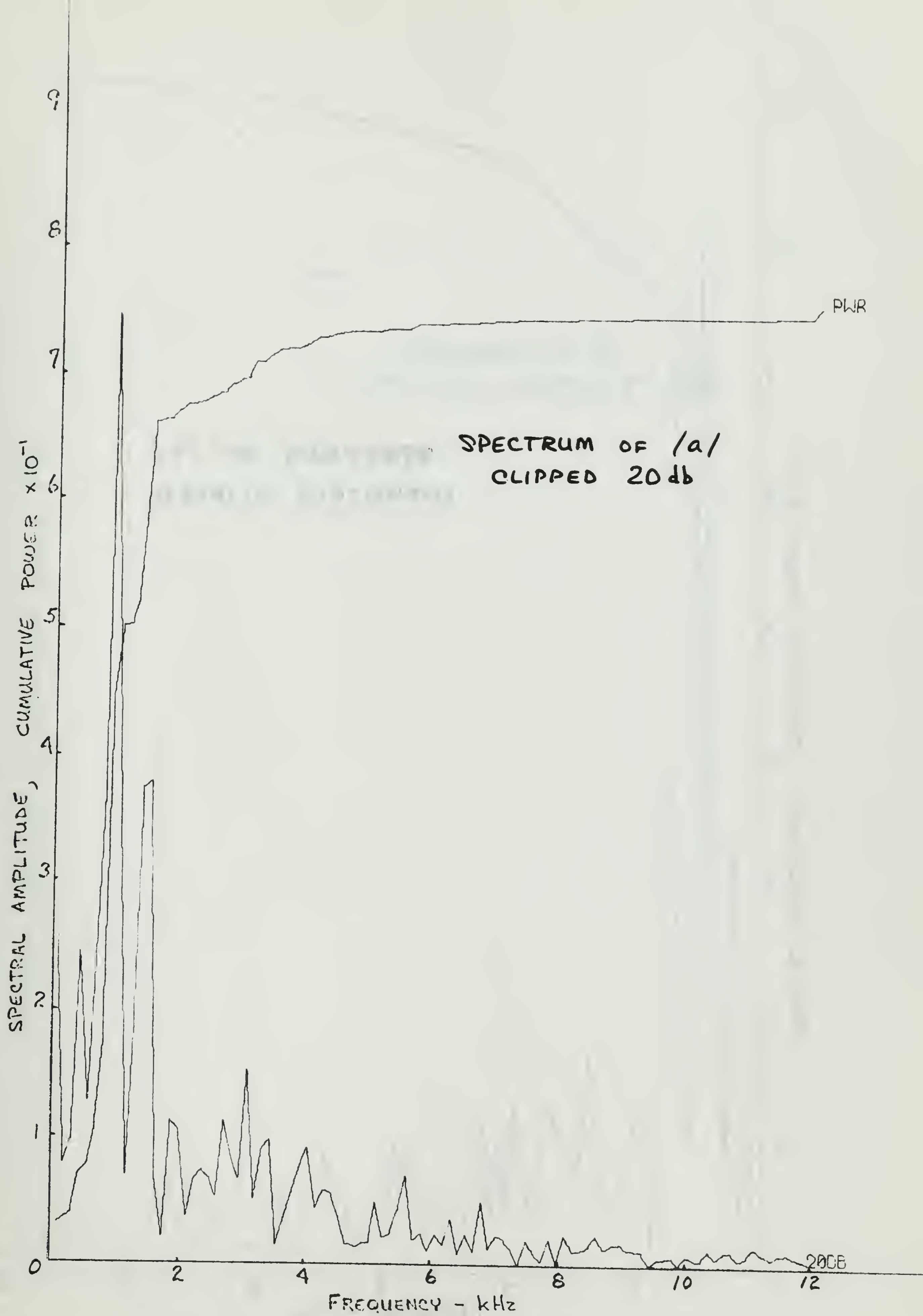

SPECTRUM OF "BED"
GRADUALLY CLIPPED 40 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER $\times 10^{-1}$
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

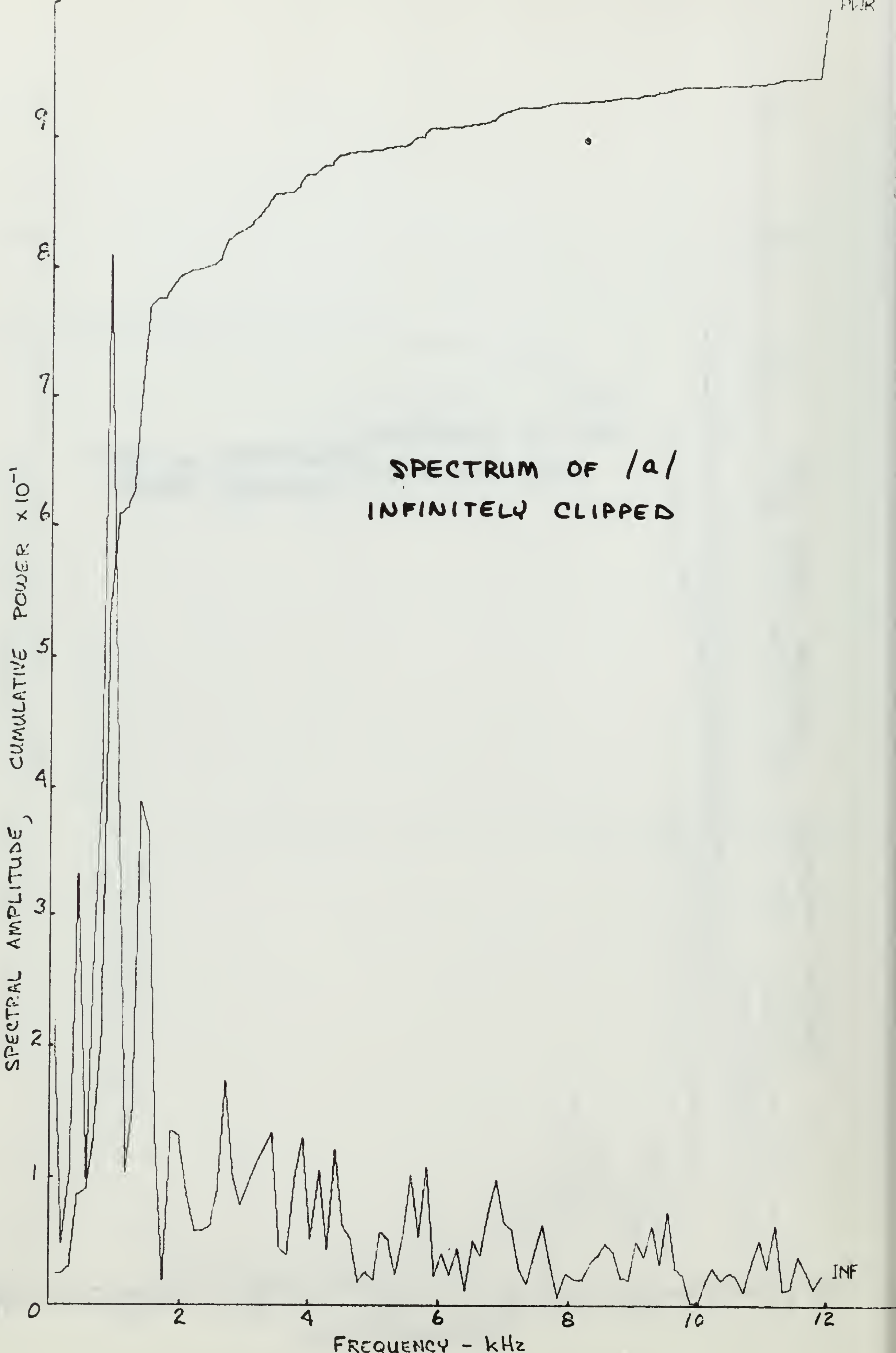


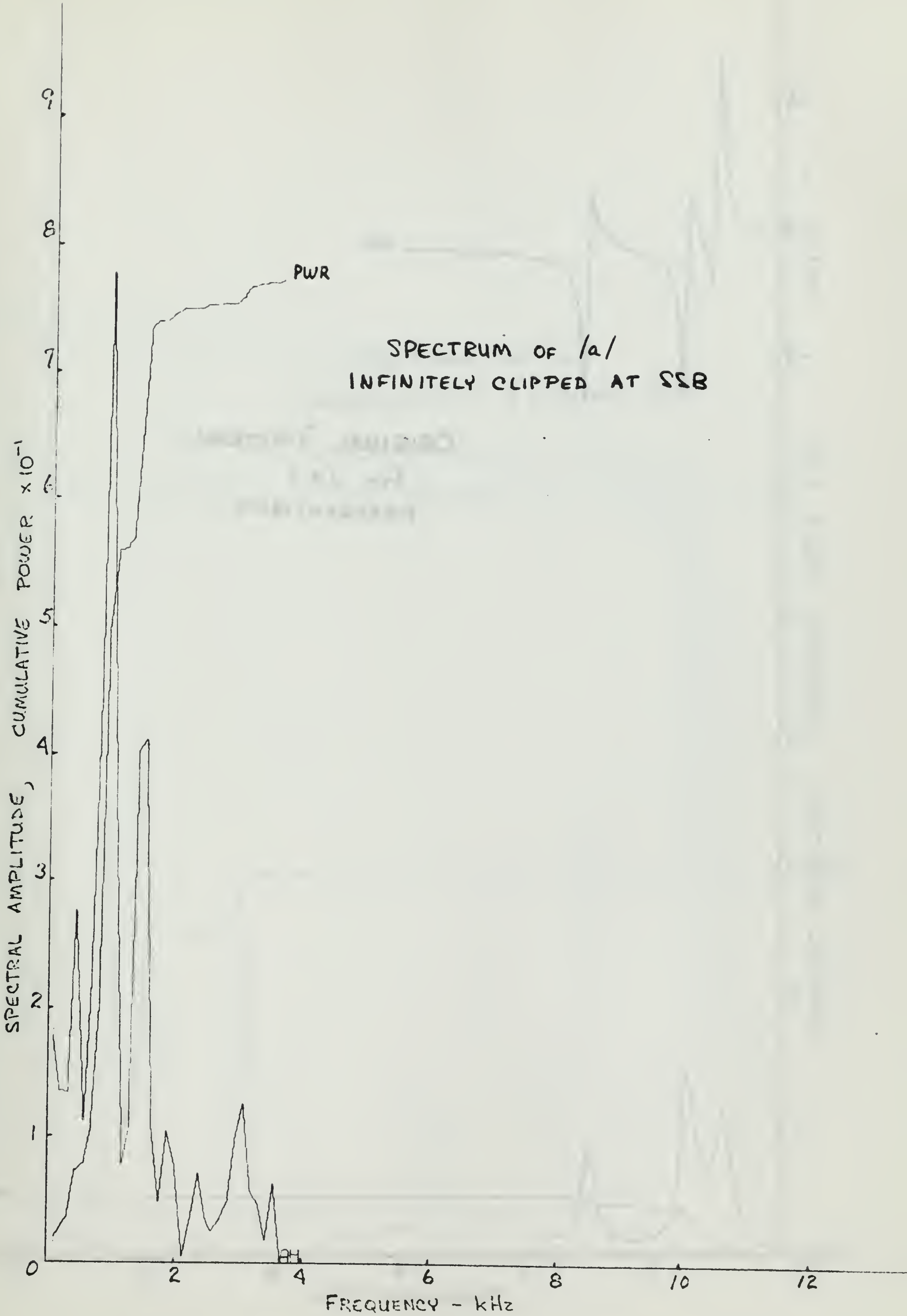
SPECTRUM OF "BED"
GRADUALLY CLIPPED 40db AT SSB
PWR
BED
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
9
8
7
6
5
4
3
2
1
0
2
4
6
8
10
12
FREQUENCY - kHz

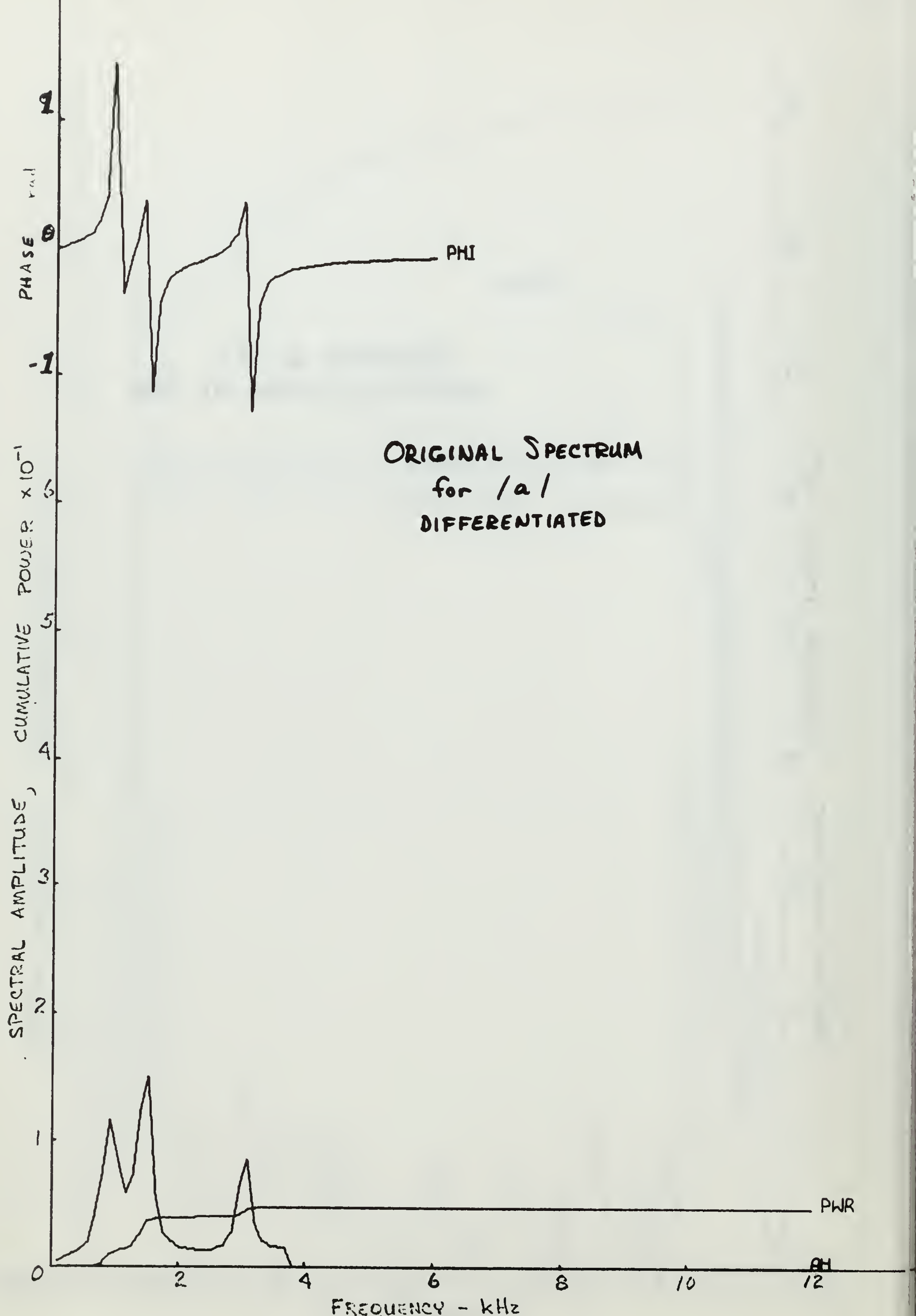

PHI
ORIGINAL SPECTRUM
for /a/
SPECTRAL AMPLITUDE, CUMULATIVE POWER $\times 10^{-1}$
PWR
AH
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

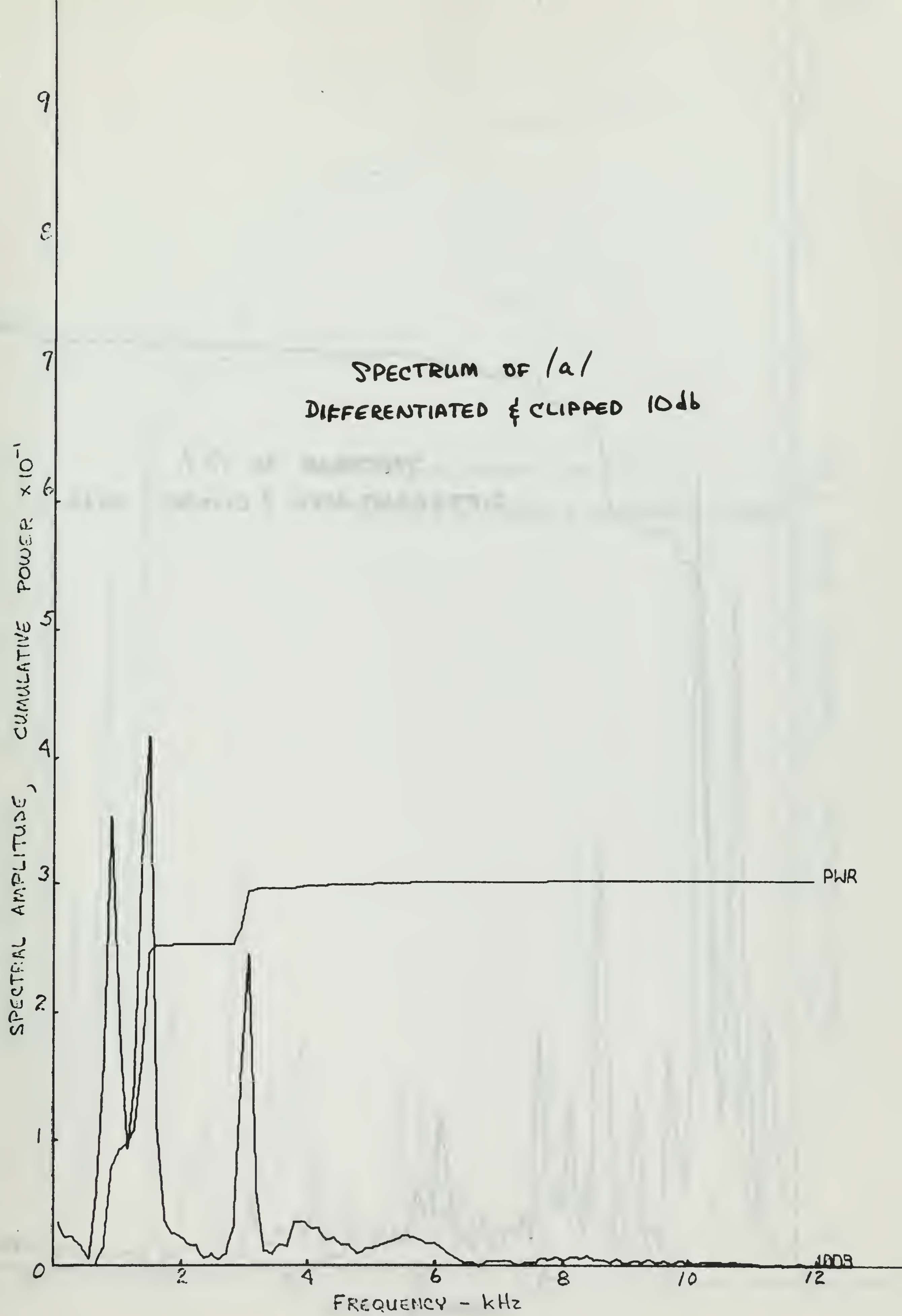

SPECTRUM OF /a/
CLIPPED 10db
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^-1
FREQUENCY - kHz
PWR
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF /a/
CLIPPED 20db
PWR
20DB
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

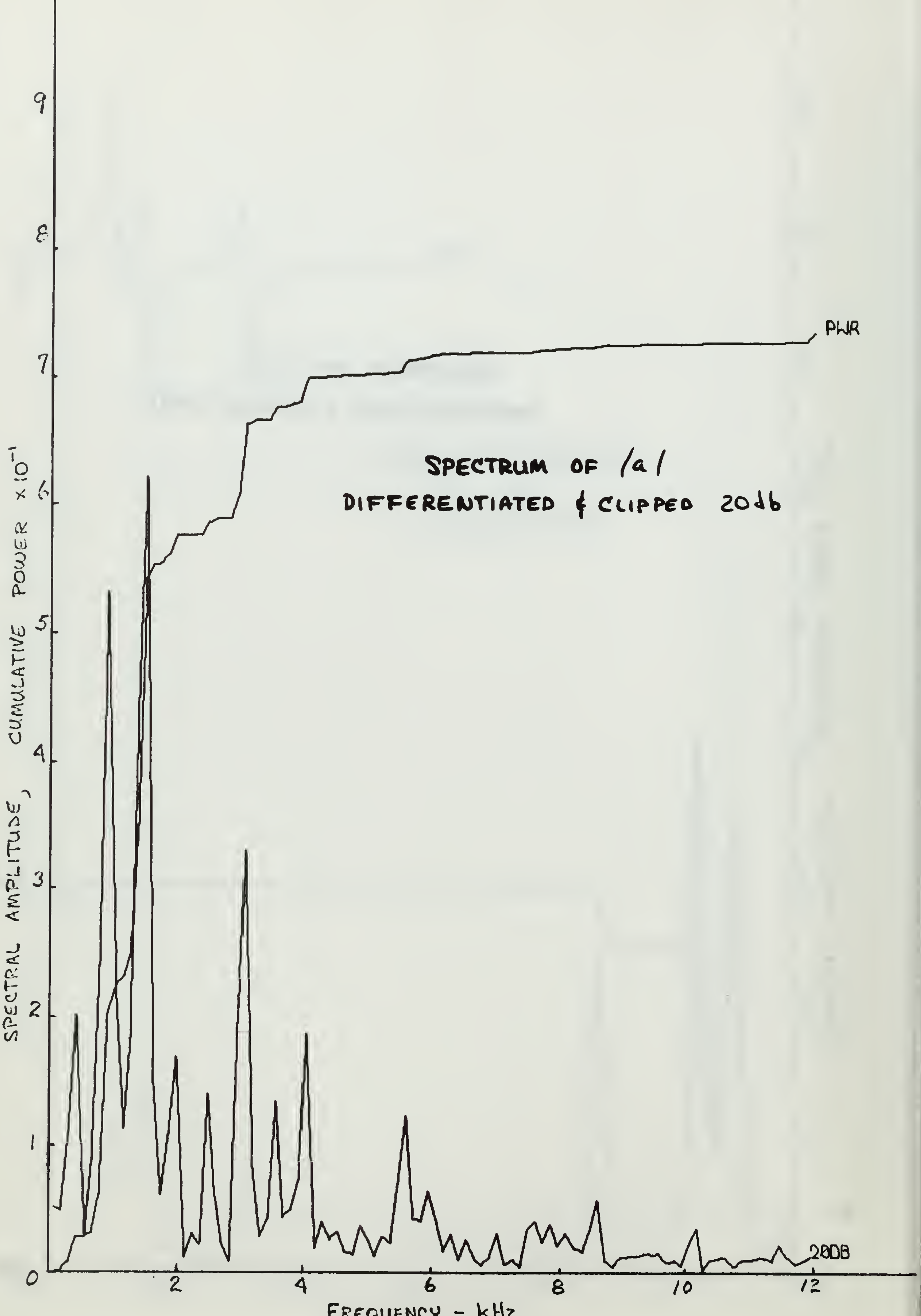

PWR
SPECTRUM OF /a/
INFINITELY CLIPPED
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
9
8
7
6
5
4
3
2
1
0
2
4
6
8
10
12
INF
FREQUENCY - kHz

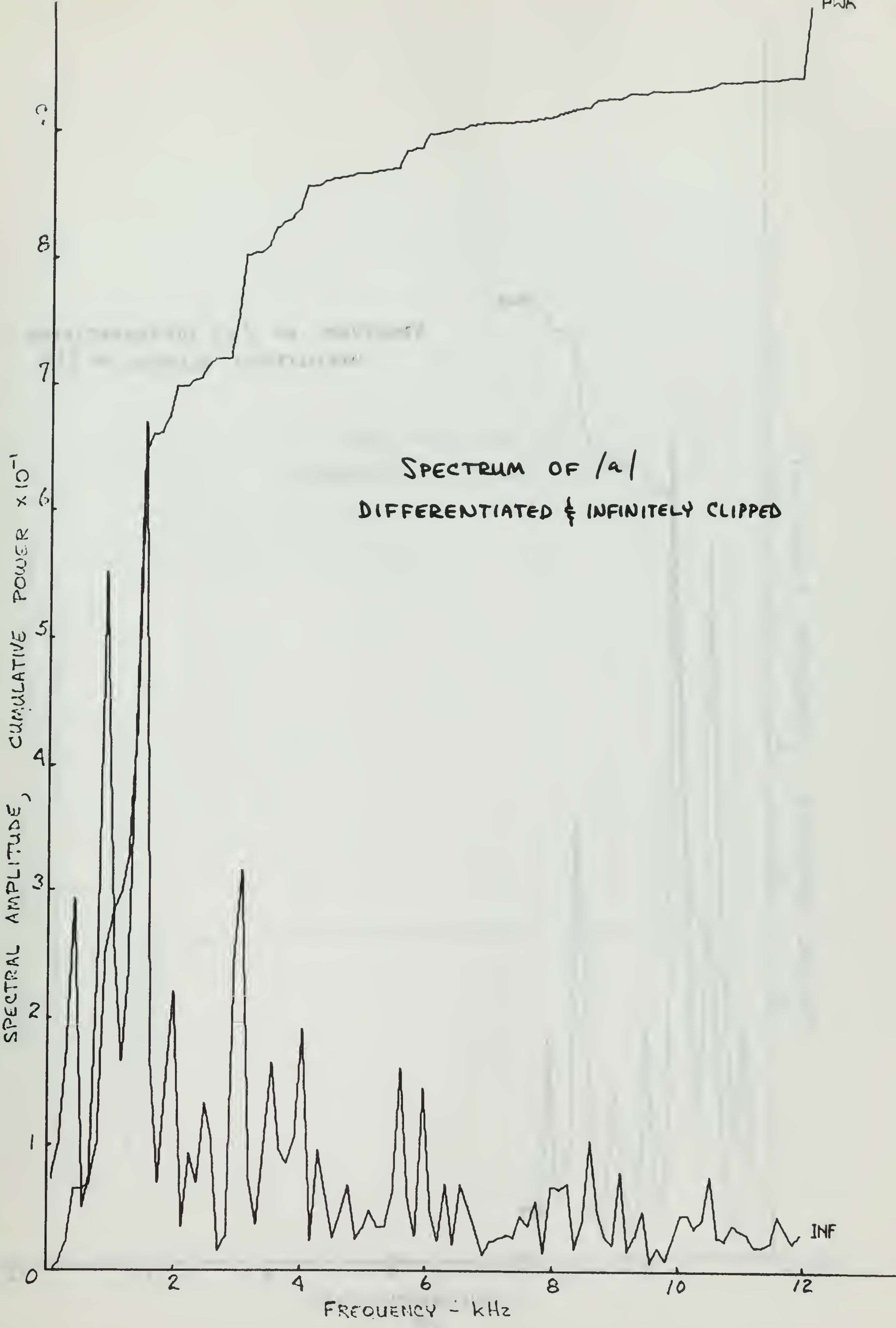

SPECTRUM OF /a/
INFINITELY CLIPPED AT SSB
PWR
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

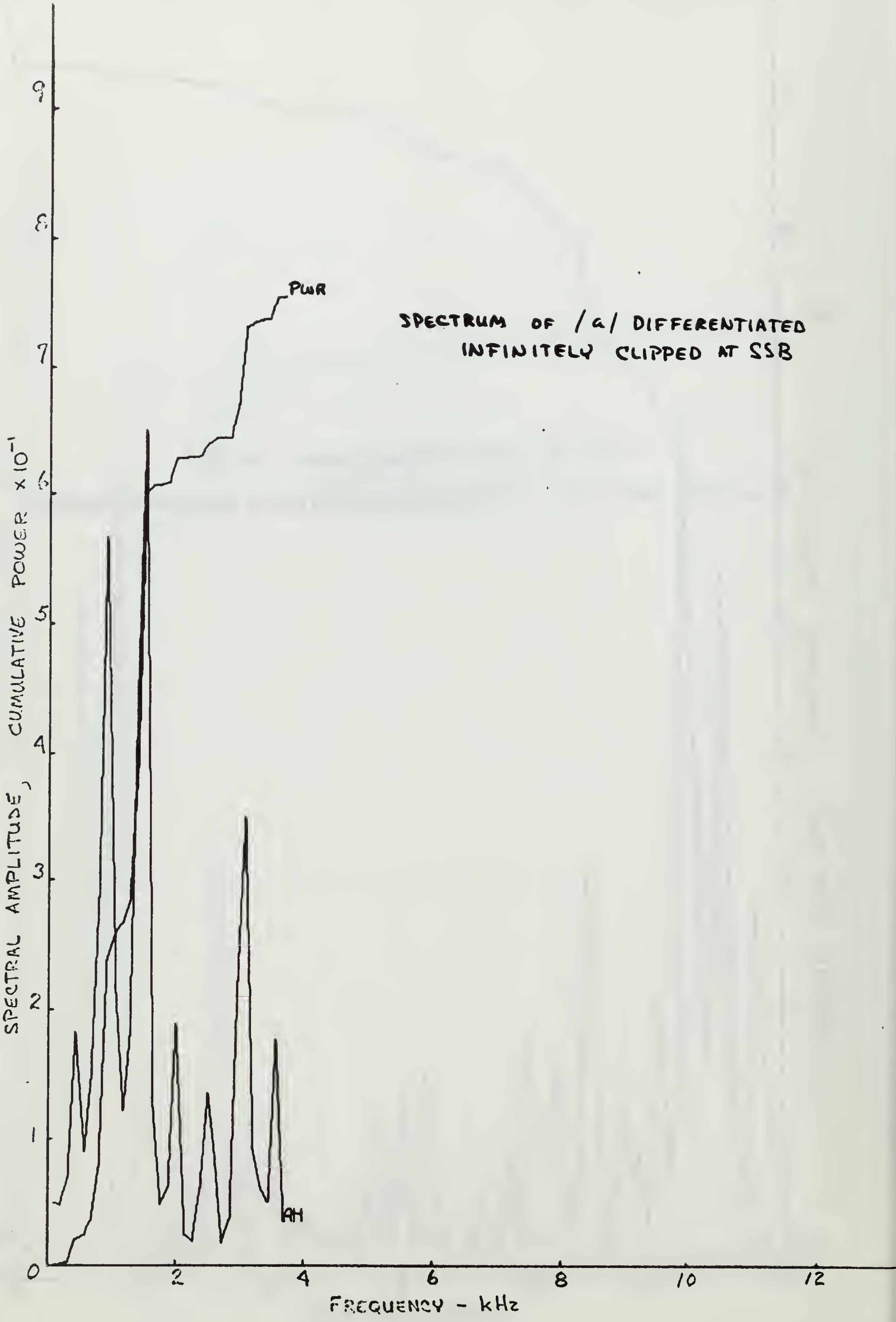

PHASE rad
1
0
-1
PHI
ORIGINAL SPECTRUM
for /a/
DIFFERENTIATED
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^-1
6
5
4
3
2
1
0
PWR
AH
2
4
6
8
10
12
FREQUENCY - kHz



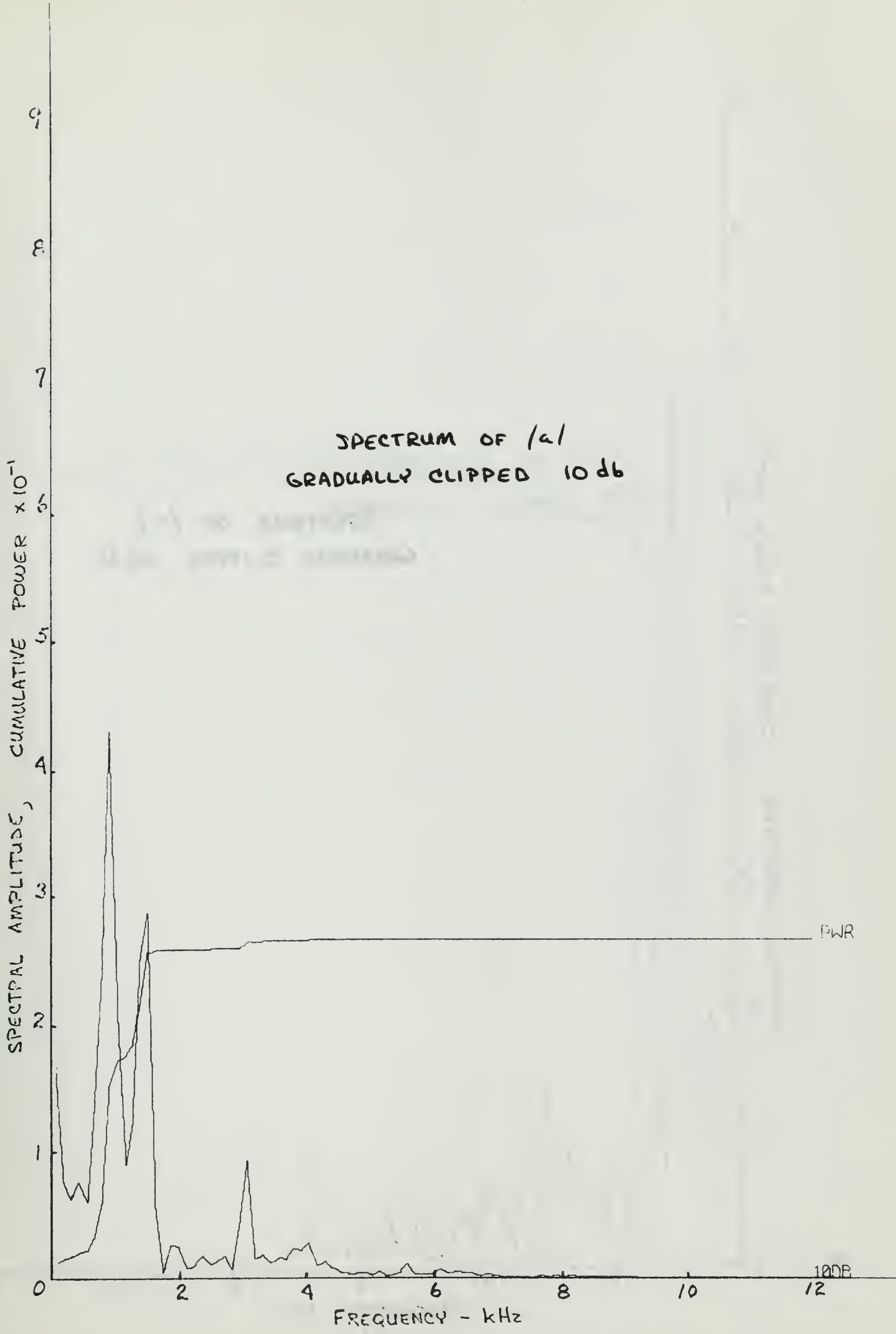
SPECTRUM OF /a/
DIFFERENTIATED & CLIPPED 10db
SPECTRAL AMPLITUDE, CUMULATIVE POWER $\times 10^{-1}$
9
8
7
6
5
4
3
2
1
0
PWR
2
4
6
8
10
12
FREQUENCY - kHz

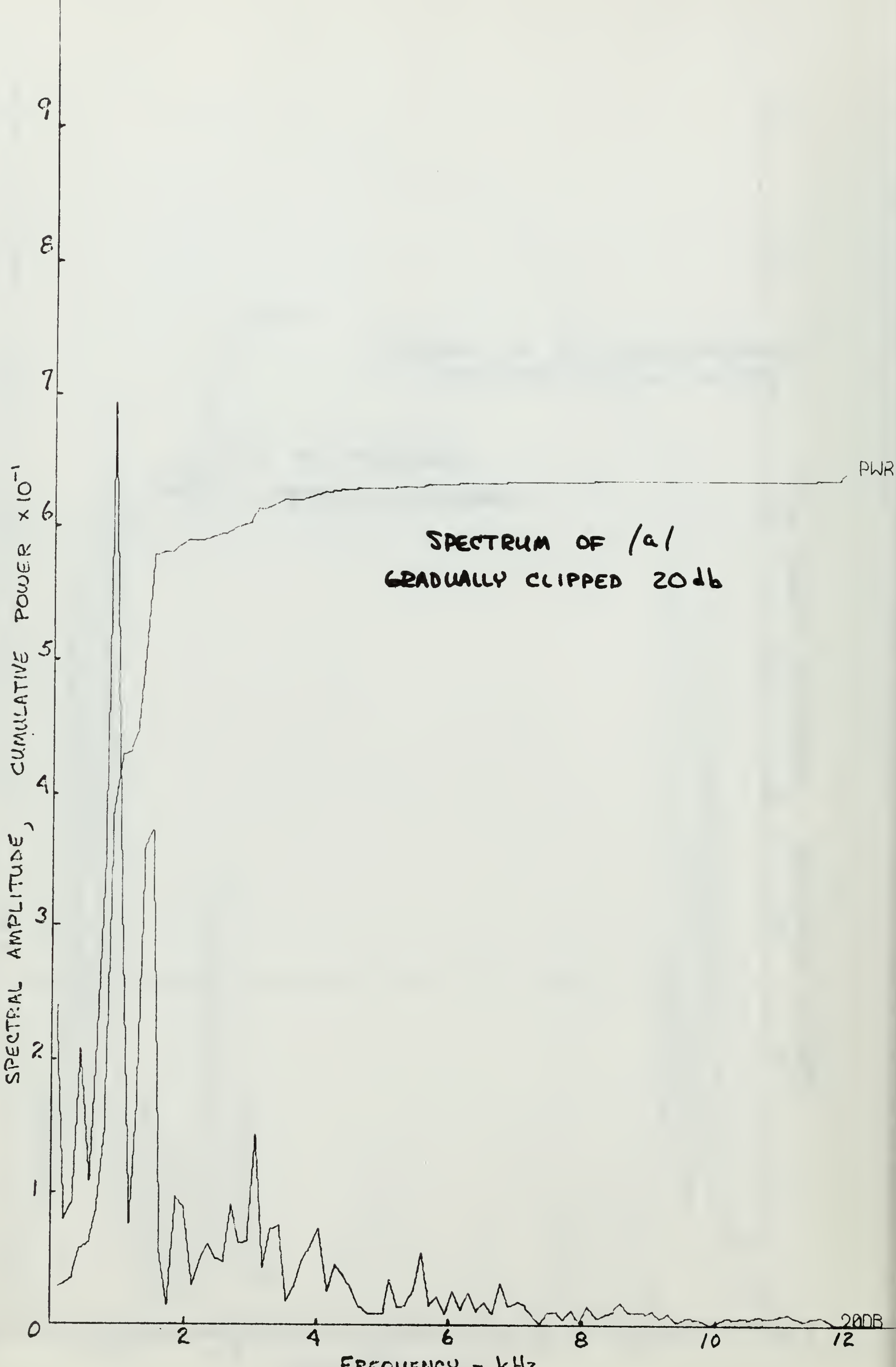

SPECTRUM OF /a/
DIFFERENTIATED & CLIPPED 20db
PWR
20DB
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF /a/
DIFFERENTIATED & INFINITELY CLIPPED
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^-1
FREQUENCY - kHz
INF
0
1
2
3
4
5
6
7
8
2
4
6
8
10
12

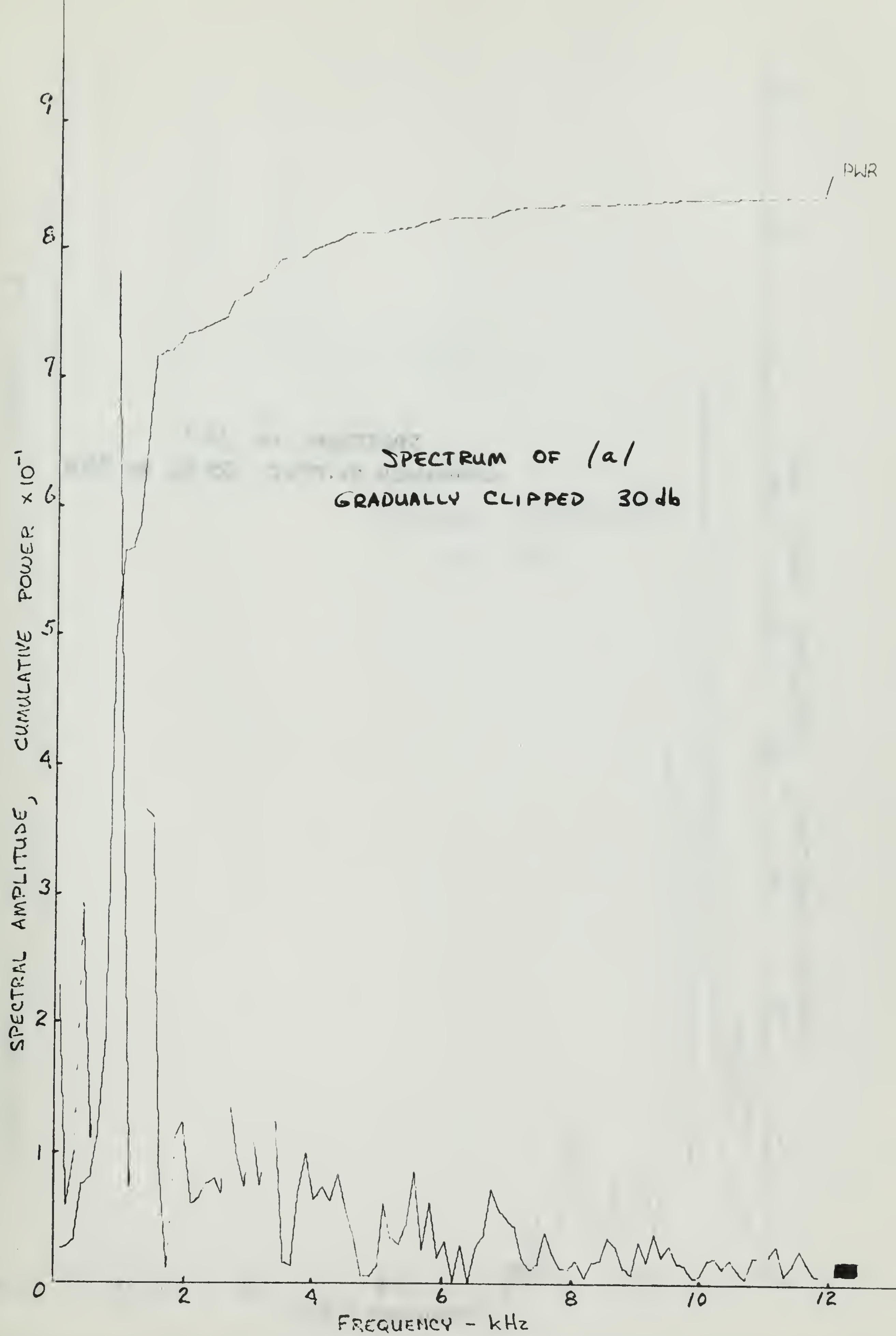


SPECTRUM OF /a/ DIFFERENTIATED INFINITELY CLIPPED AT SSB
PWR
AH
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

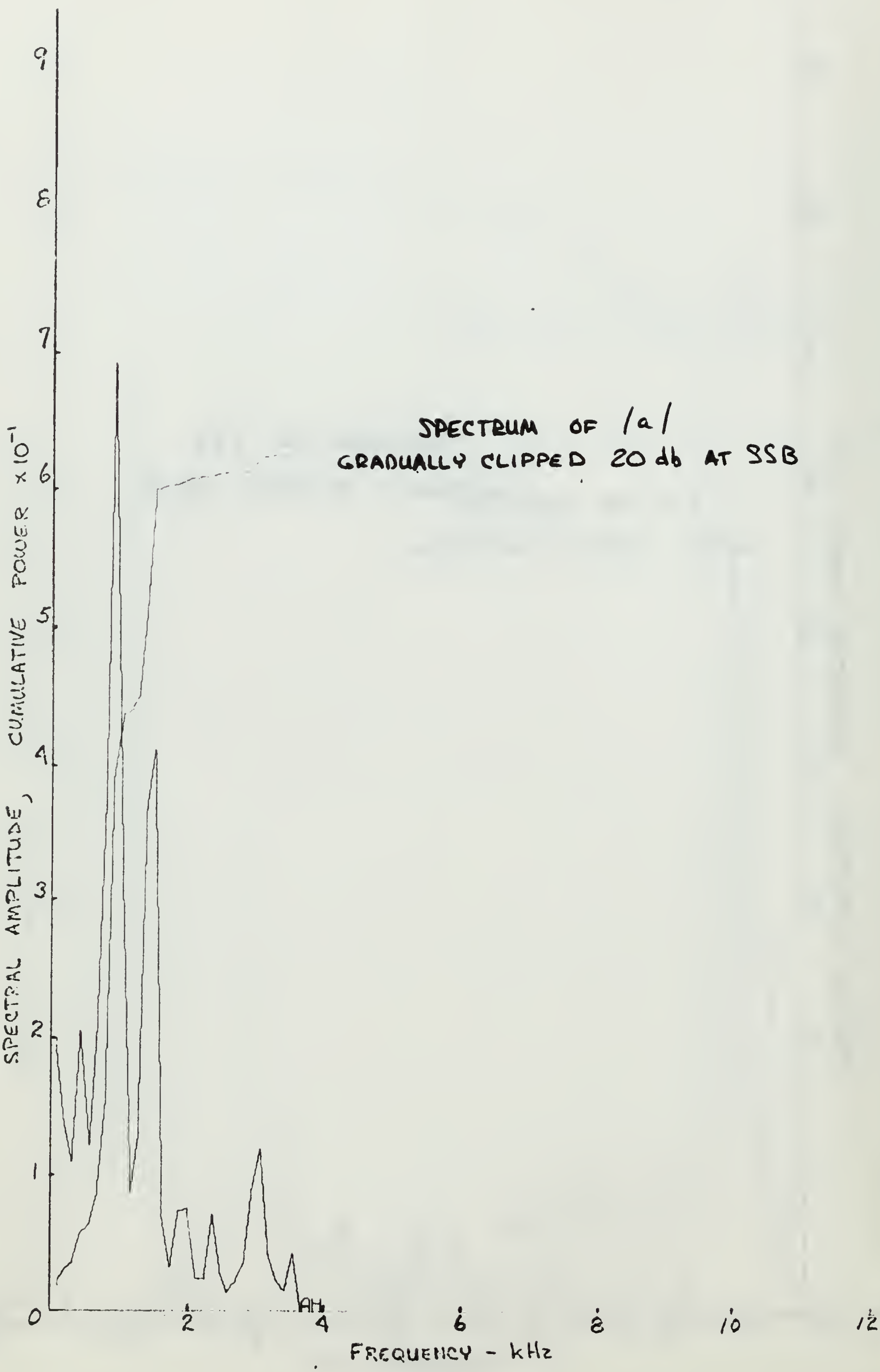

SPECTRUM OF /a/
GRADUALLY CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^{-1}
PWR
10DB
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

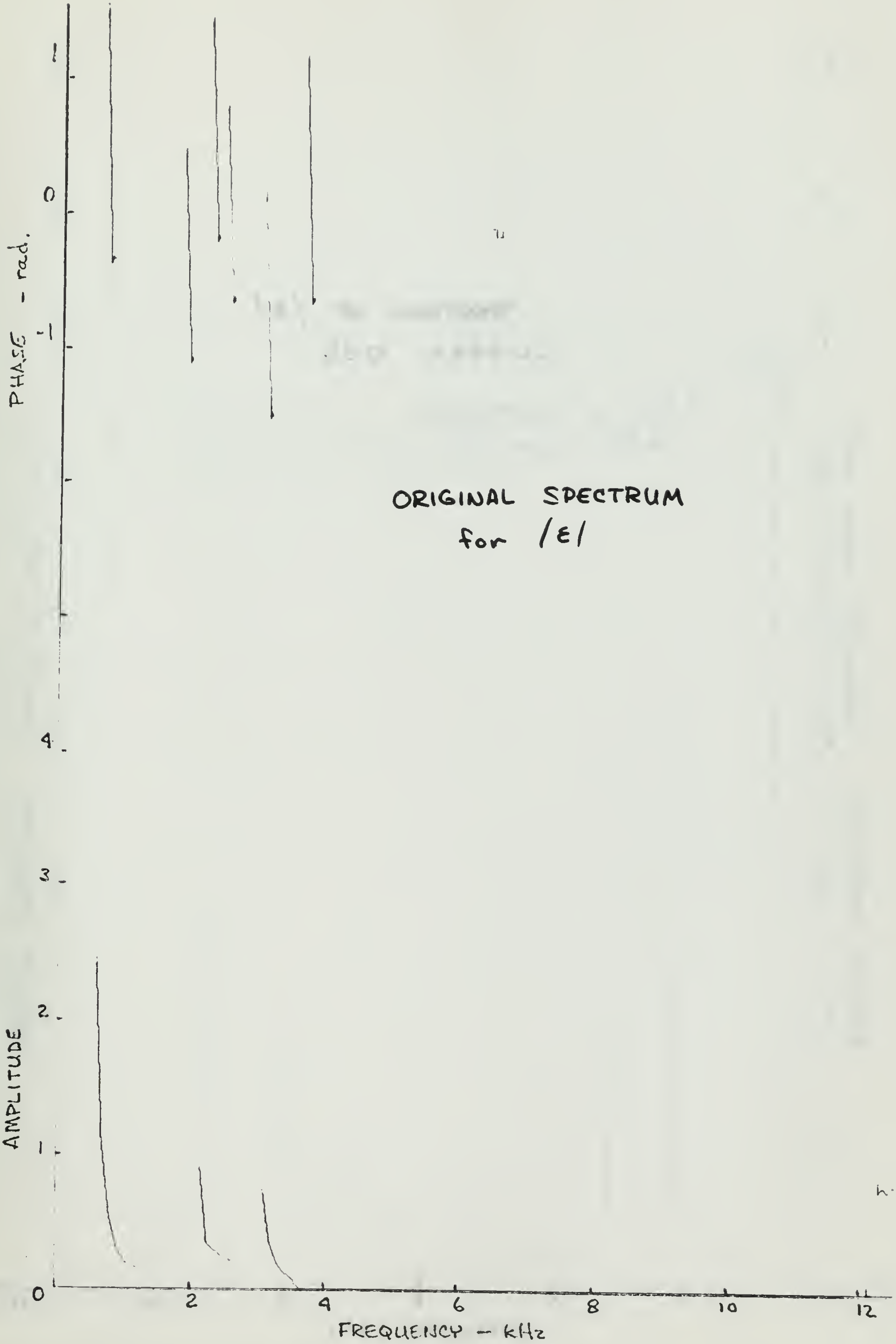


SPECTRUM OF /a/
GRADUALLY CLIPPED 20db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
PWR
20DB
0
1
2
3
4
5
6
7
8
9
10
12

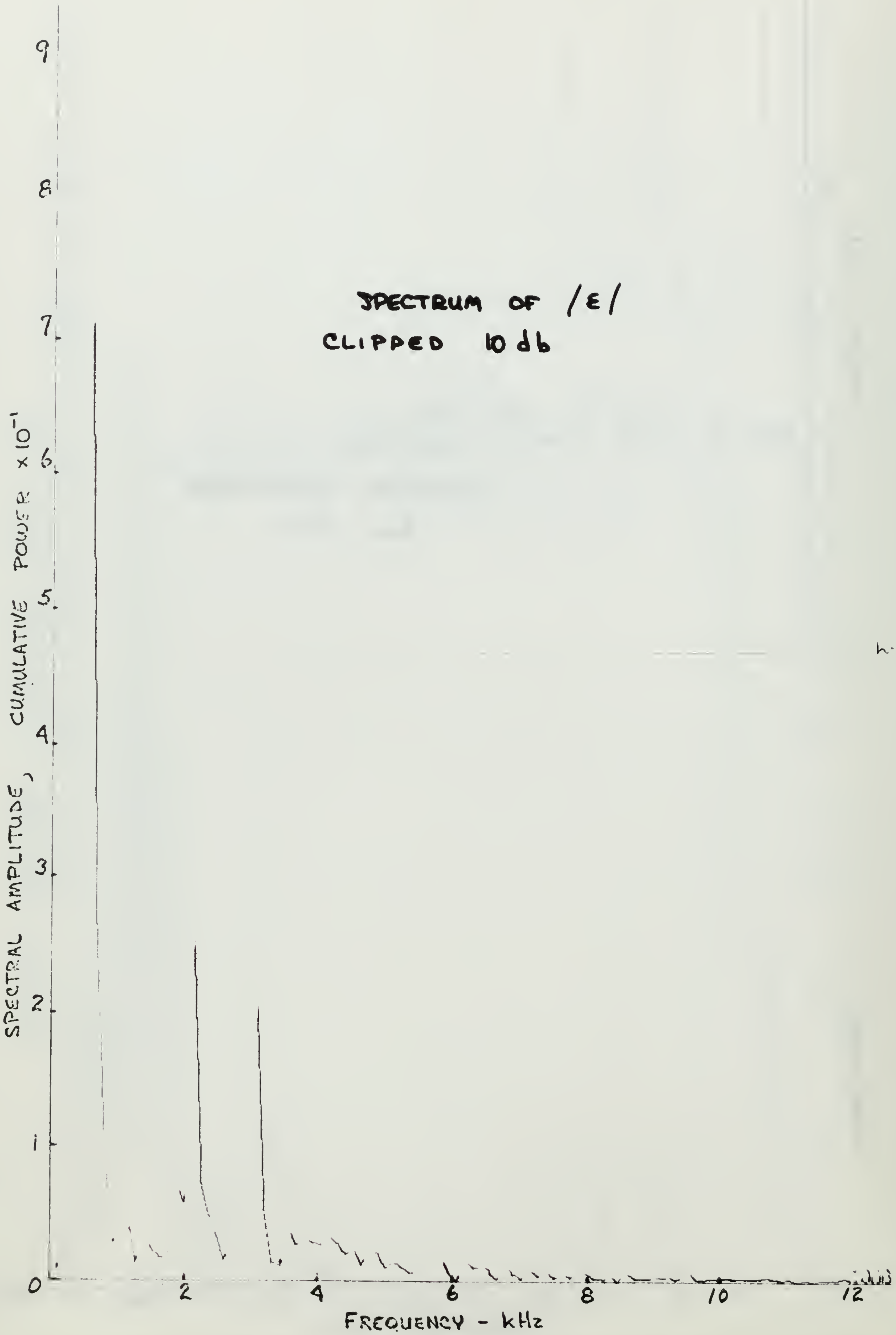


PWR
SPECTRUM OF /a/
GRADUALLY CLIPPED 30 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12
FREQUENCY - kHz

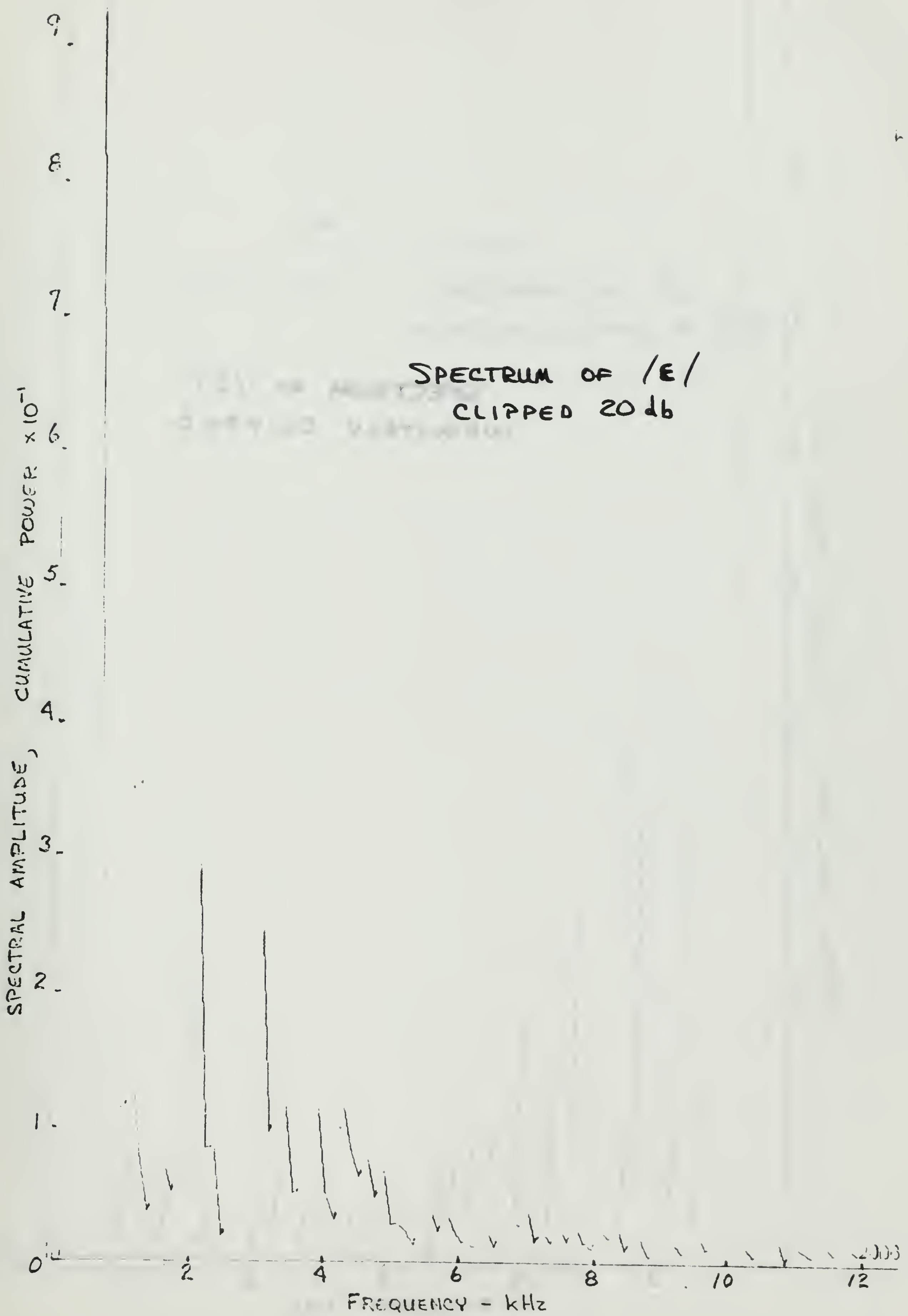

SPECTRUM OF /a/
GRADUALLY CLIPPED 20 db AT SSB
SPECTRAL AMPLITUDE, CUMULATIVE POWER $\times 10^{-1}$
9
8
7
6
5
4
3
2
1
0
2
4
6
8
10
12
FREQUENCY - kHz



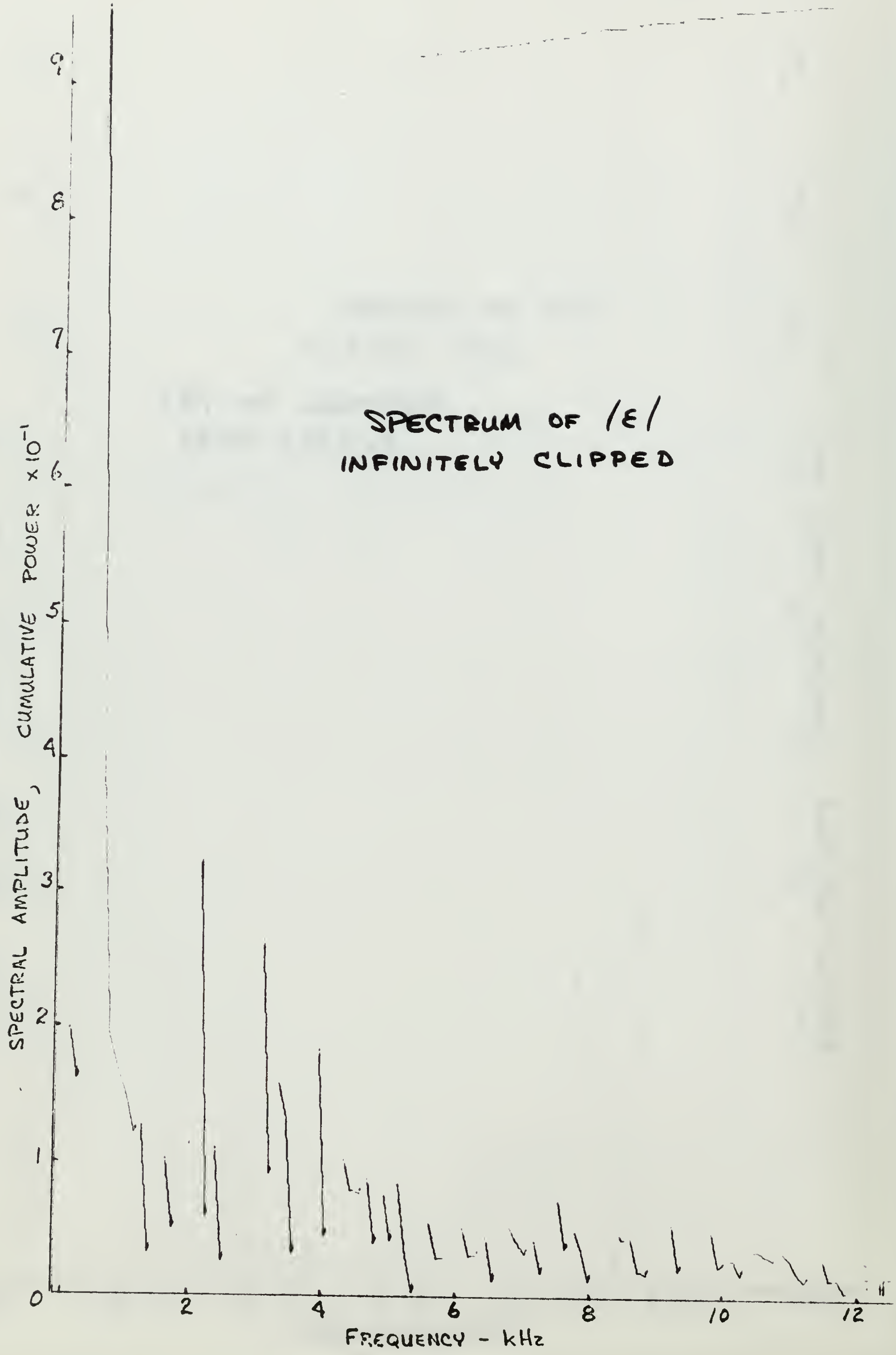
PHASE - rad.
1
0
-1
ORIGINAL SPECTRUM
for /ε/
AMPLITUDE
4
3
2
1
0
2
4
6
8
10
12
FREQUENCY - kHz



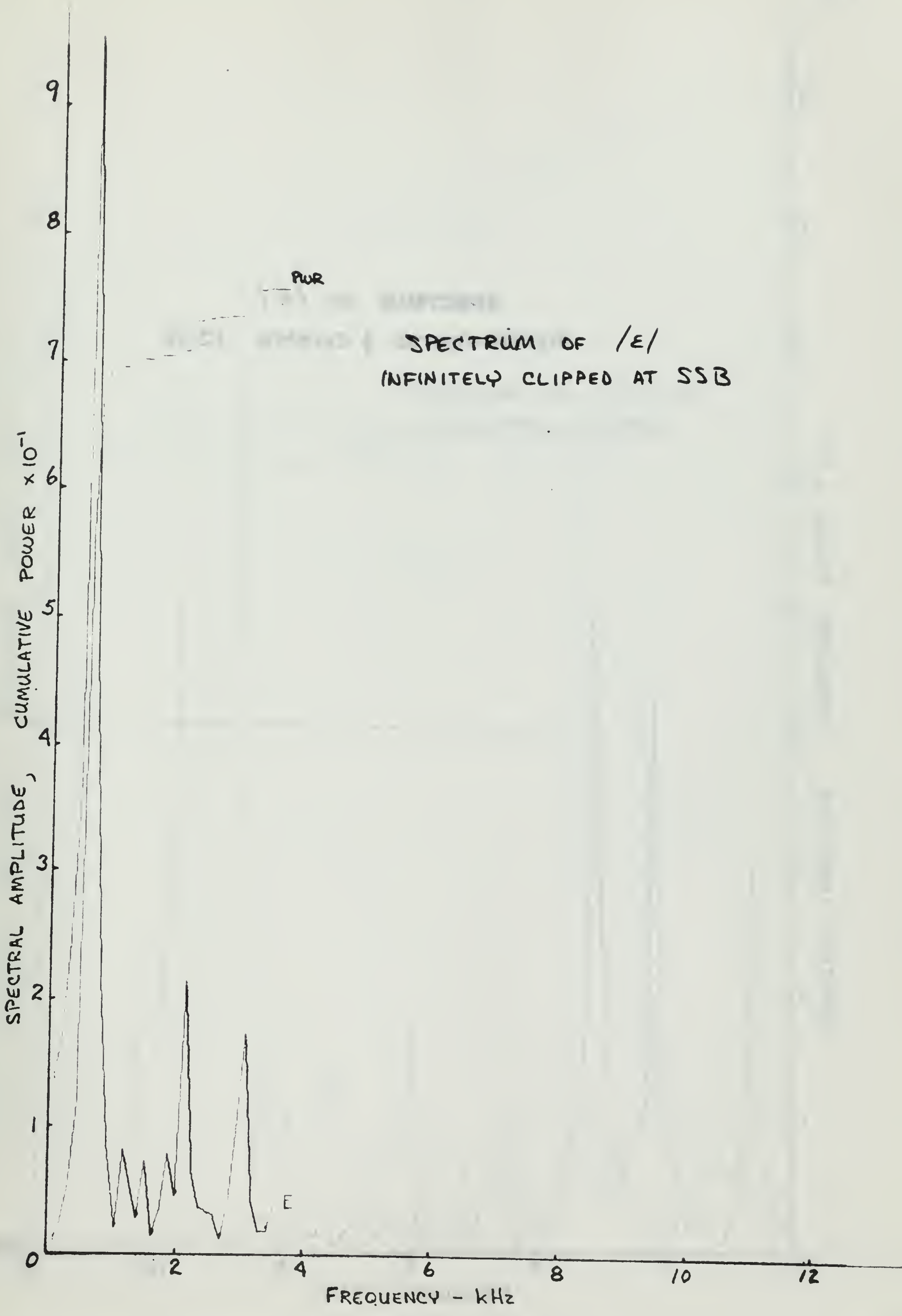
SPECTRUM OF /ε/
CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12
FREQUENCY - kHz



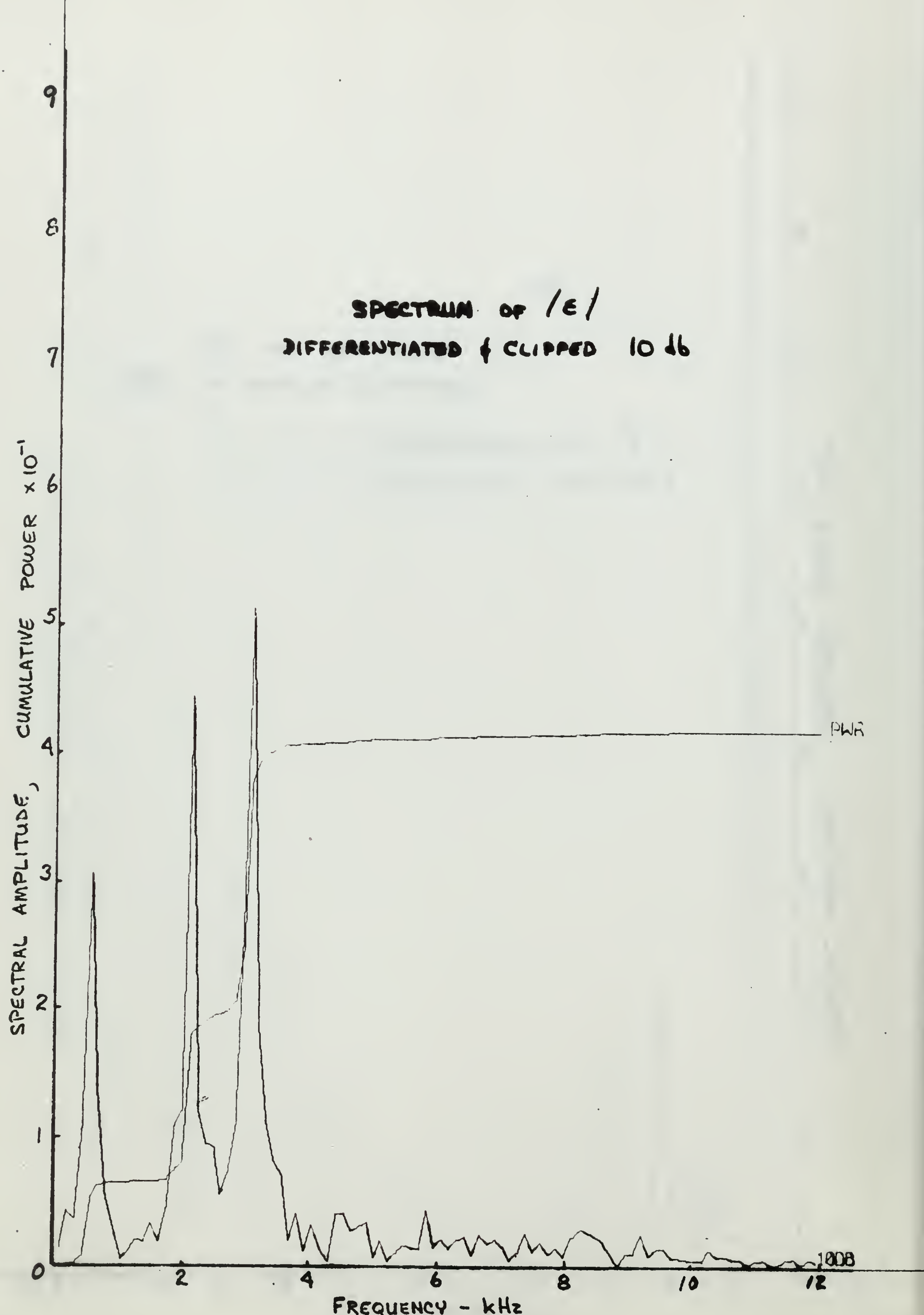
SPECTRUM OF /ɛ/
CLIPPED 20 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

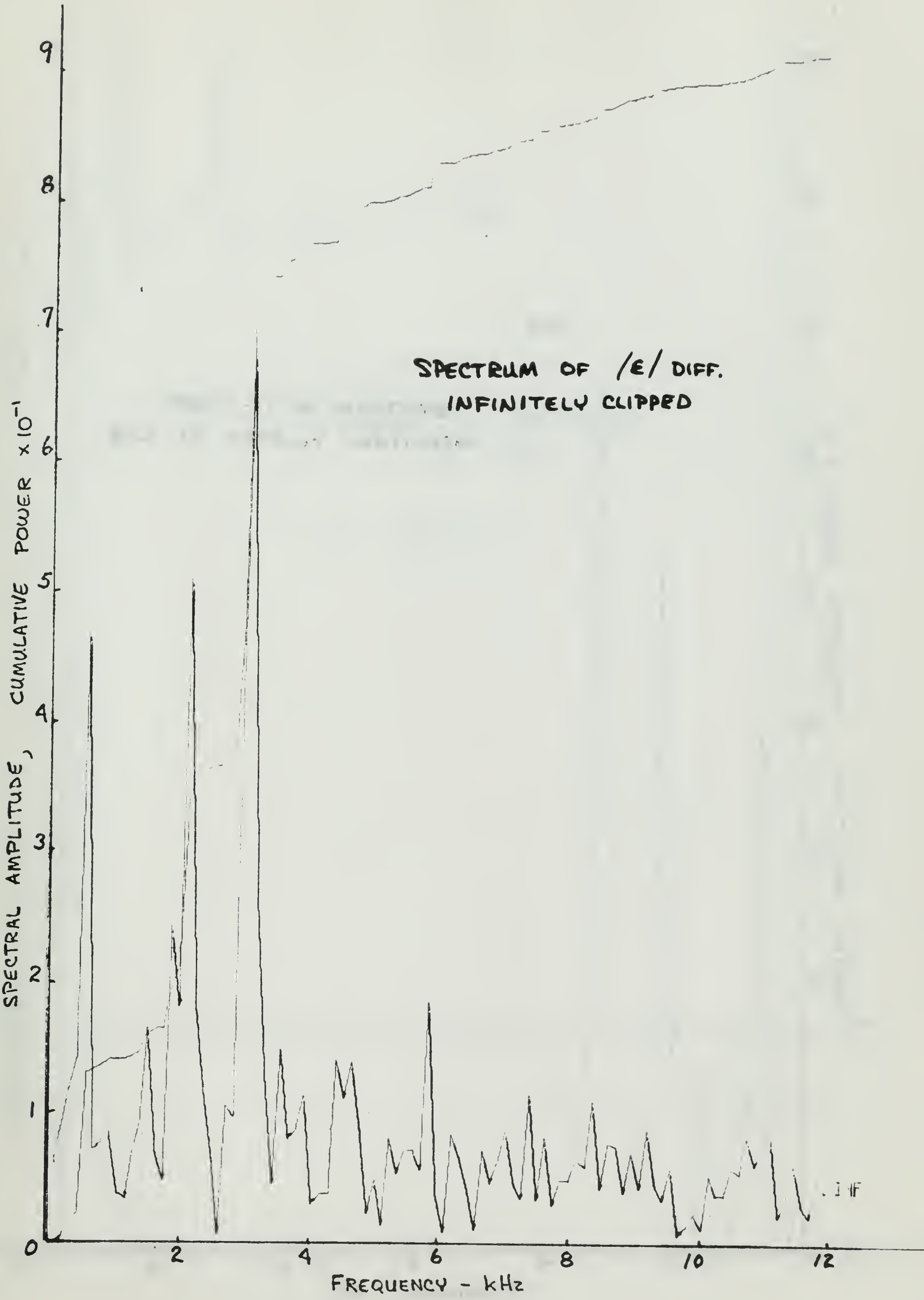


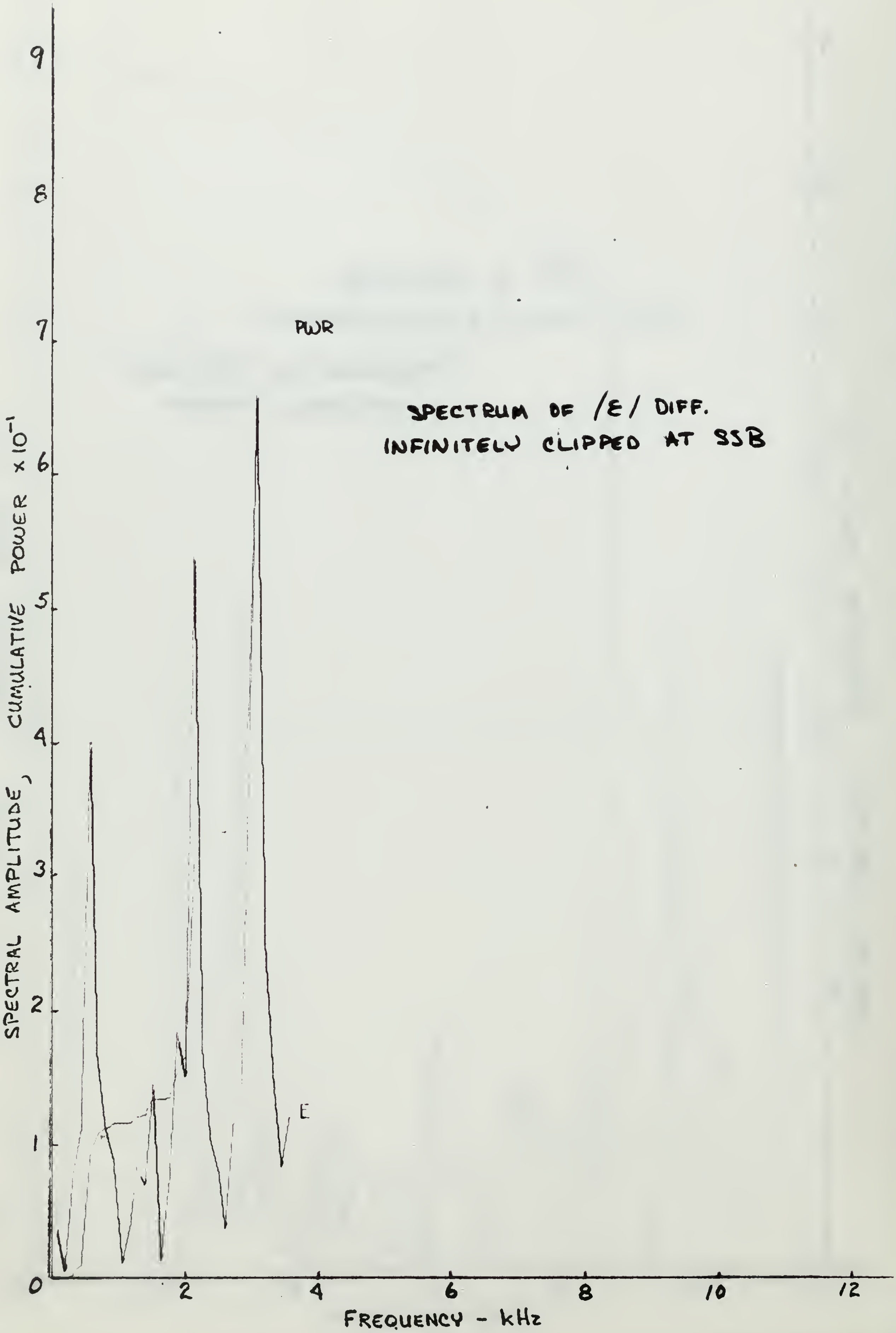
SPECTRUM OF /ɛ/
INFINITELY CLIPPED
SPECTRAL AMPLITUDE, CUMULATIVE POWER ×10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12



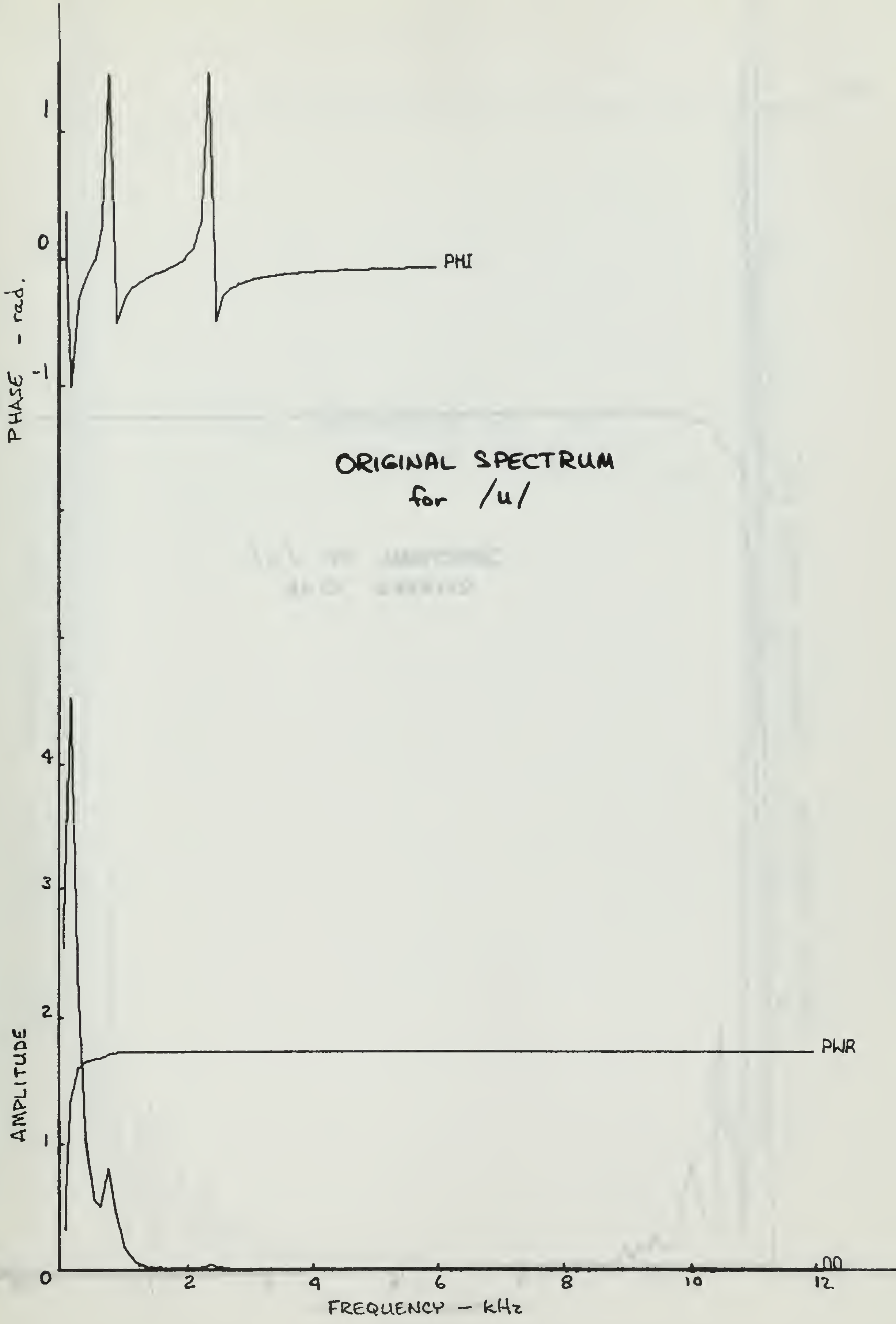
PWR
SPECTRUM OF /ε/
INFINITELY CLIPPED AT SSB
SPECTRAL AMPLITUDE, CUMULATIVE POWER × 10^{-1}
FREQUENCY - kHz
E
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF /ε/
DIFFERENTIATED & CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER × 10⁻¹
PWR
1008
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
10
12

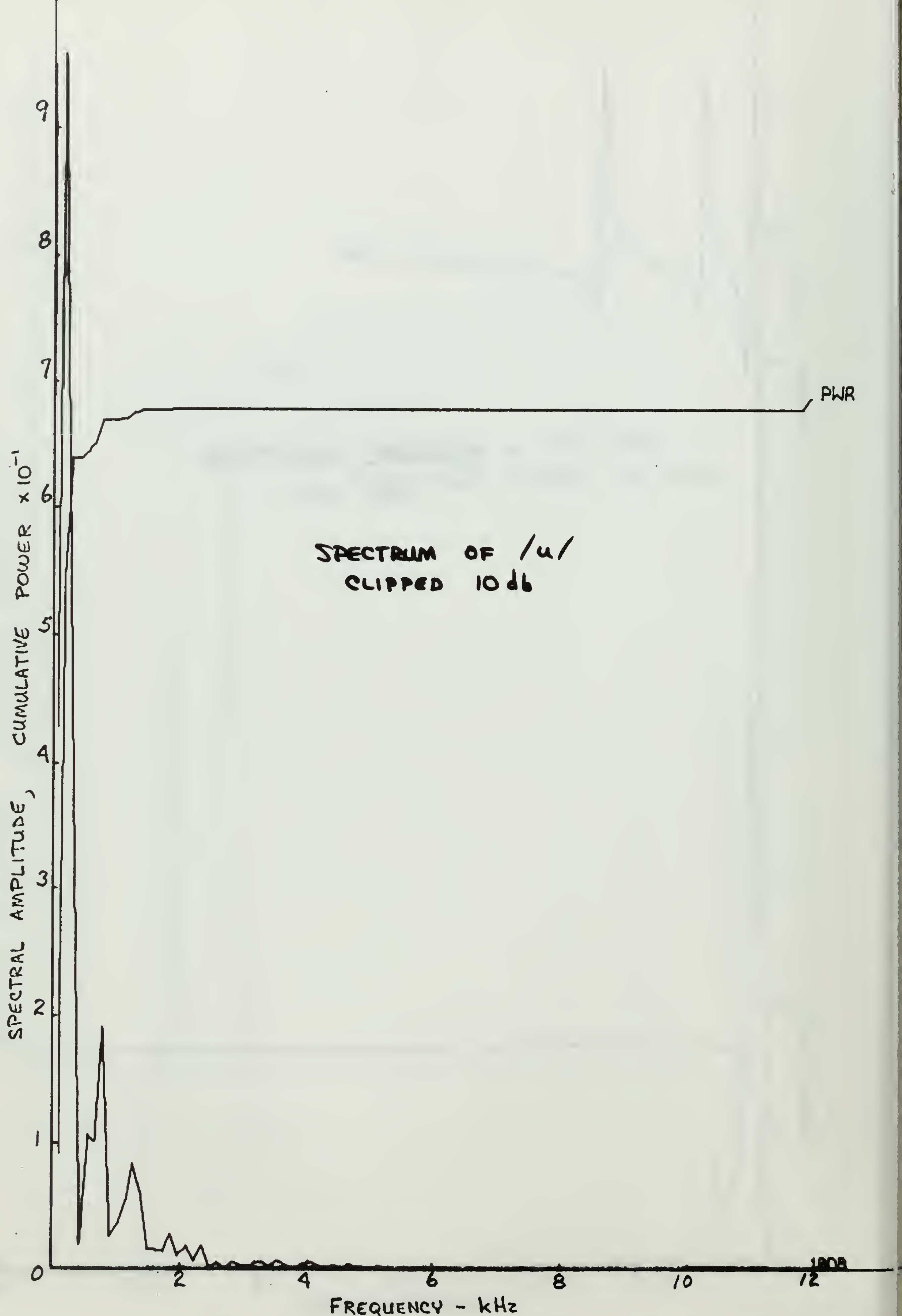

SPECTRUM OF /ɛ/ DIFF.
INFINITELY CLIPPED
SPECTRAL AMPLITUDE, CUMULATIVE POWER × 10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

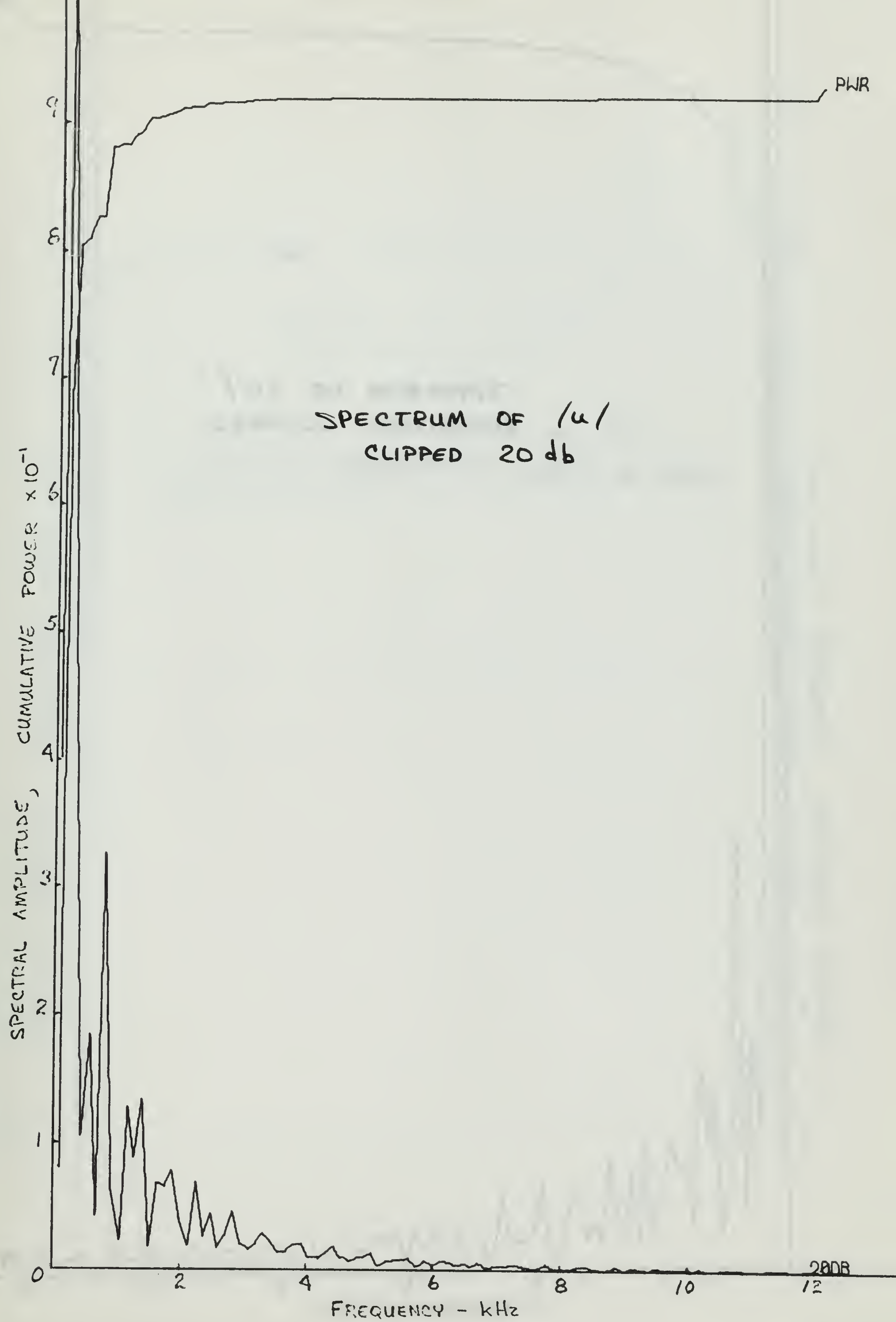


SPECTRAL AMPLITUDE, CUMULATIVE POWER × 10⁻¹
PWR
SPECTRUM OF /ε/ DIFF.
INFINITELY CLIPPED AT SSB
E
FREQUENCY - kHz

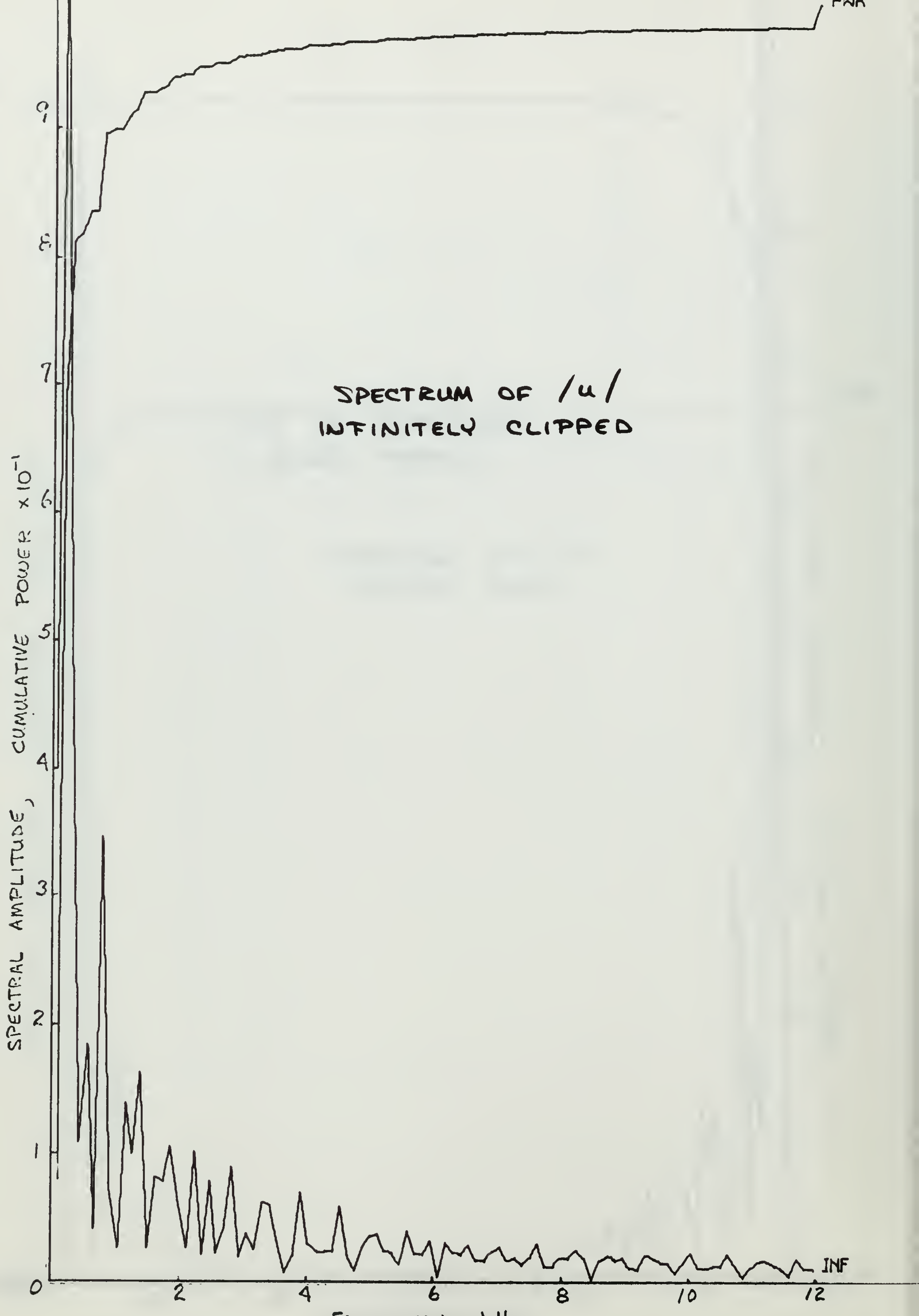


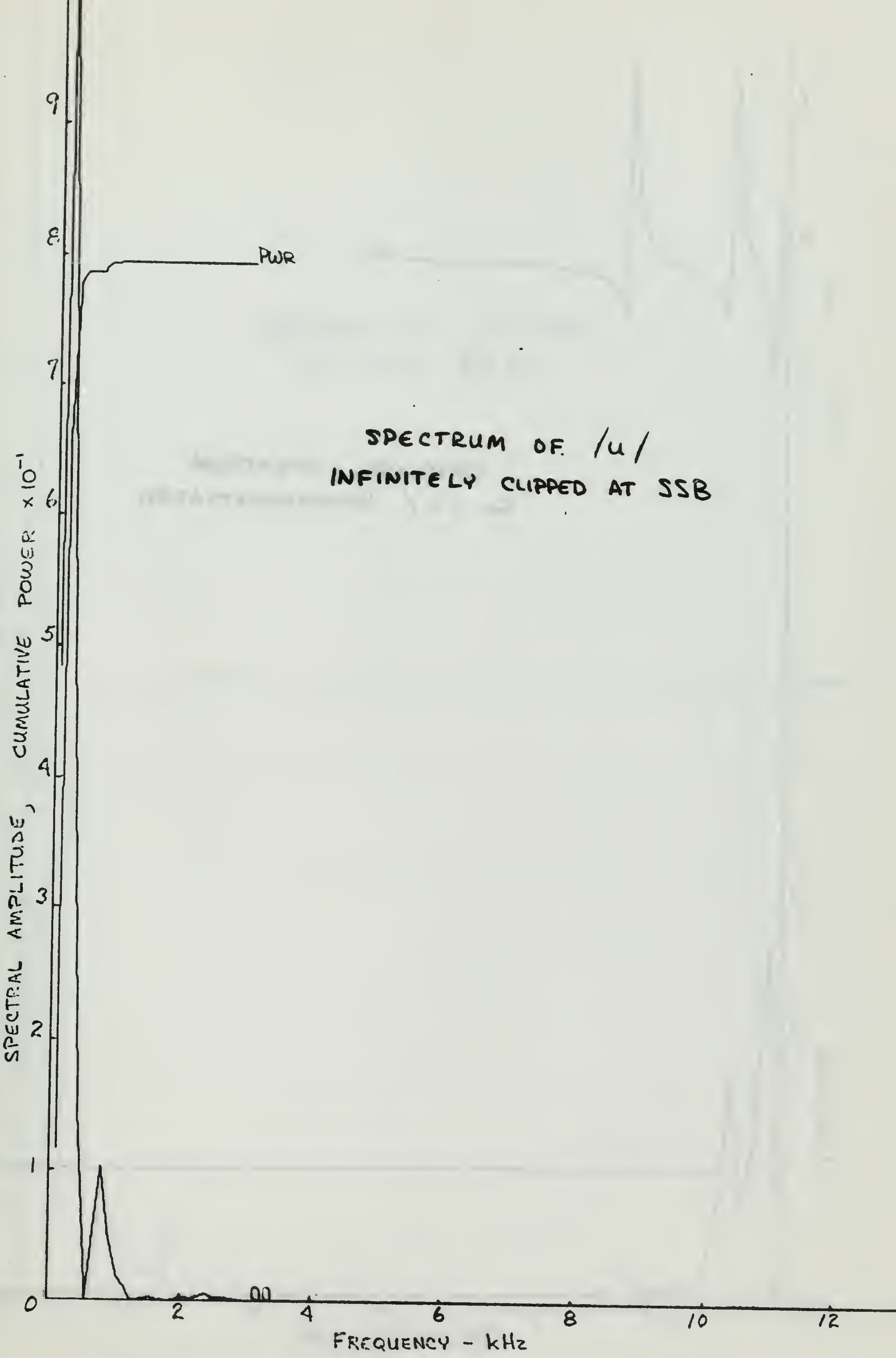
ORIGINAL SPECTRUM
for /u/


SPECTRUM OF /u/
CLIPPED 10 db
PWR
SPECTRAL AMPLITUDE, CUMULATIVE POWER x 10^-1
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
10
12

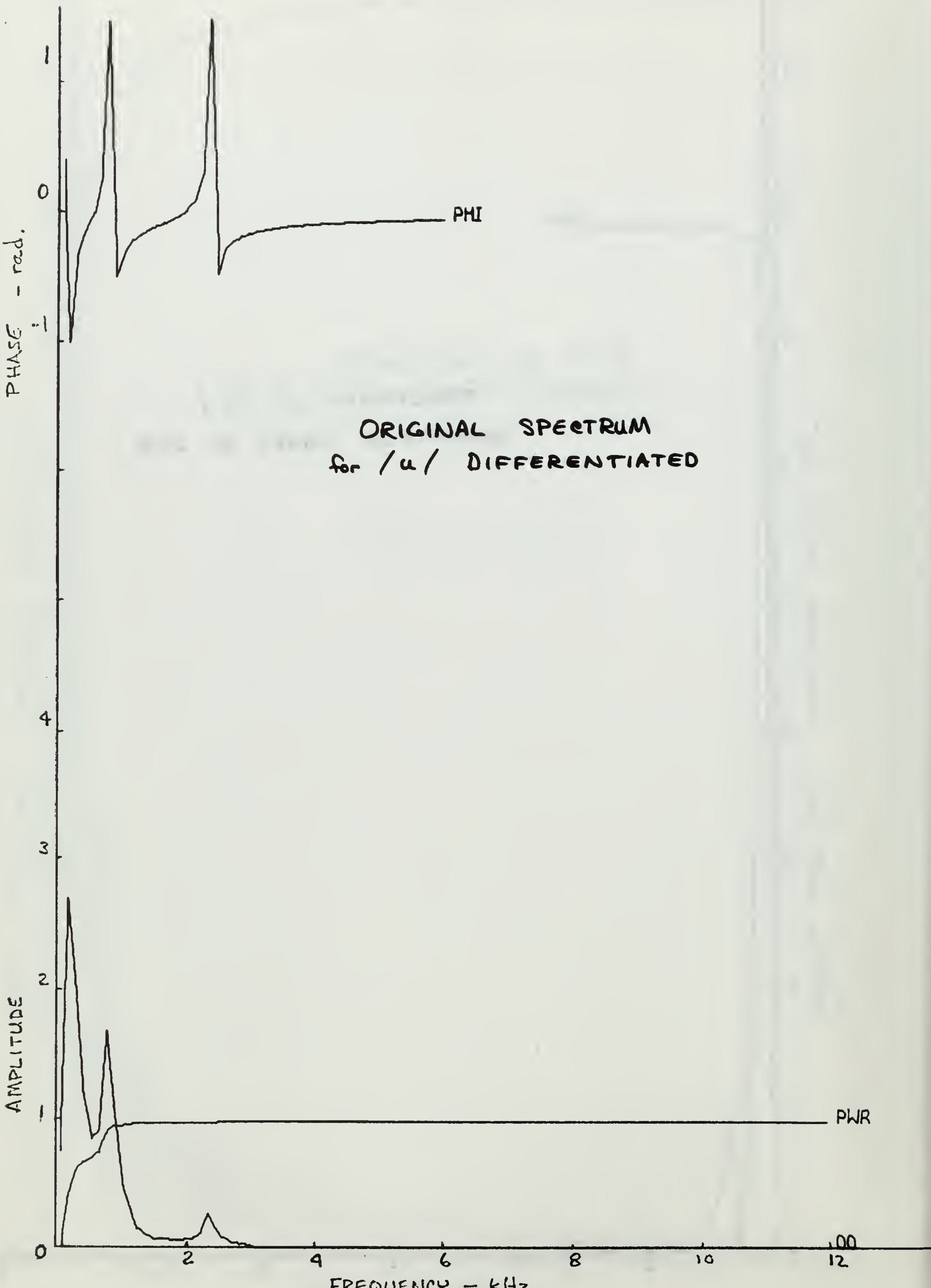


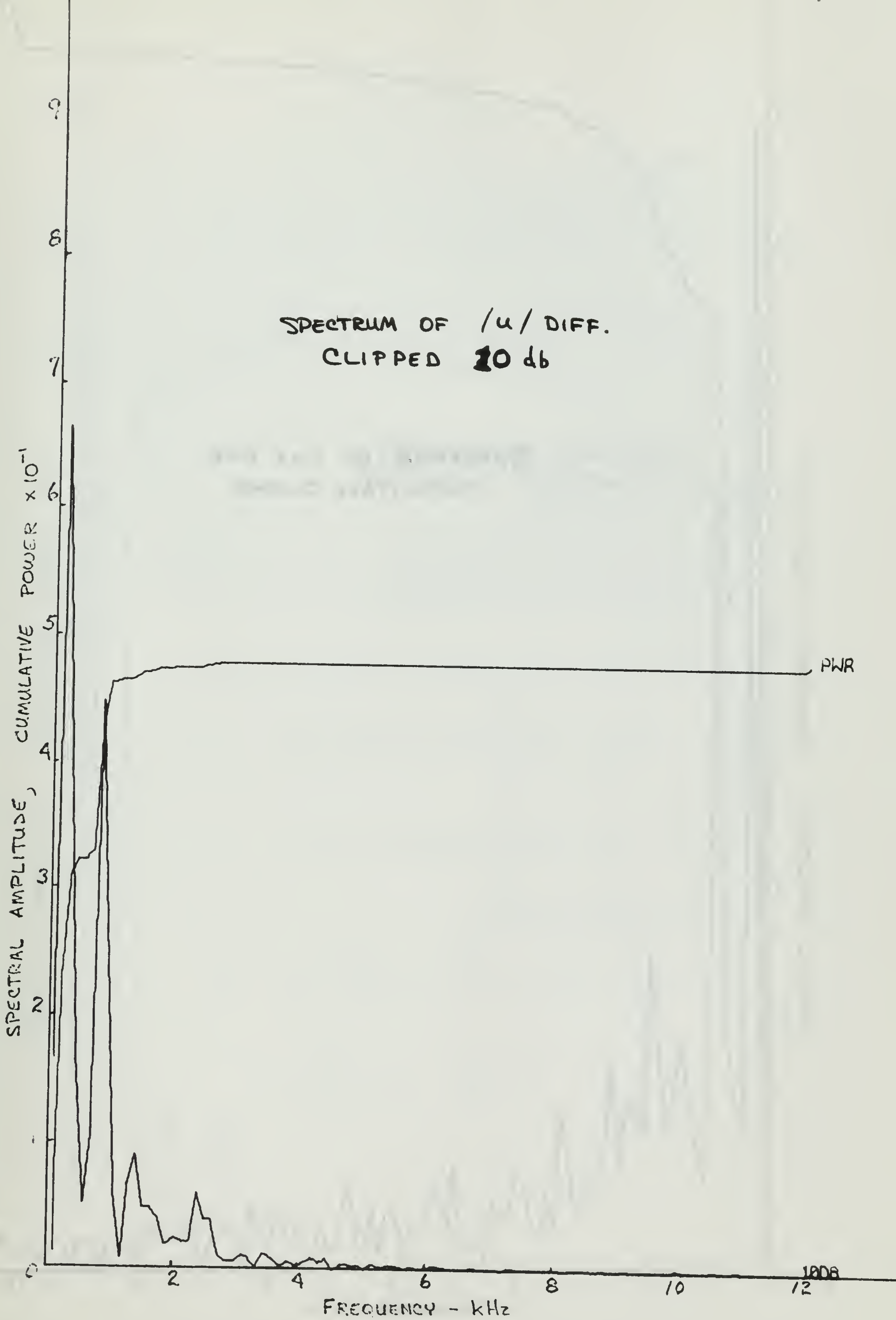
PWR
SPECTRUM OF /u/
CLIPPED 20 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER x 10^{-1}
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12
20DB
FREQUENCY - kHz

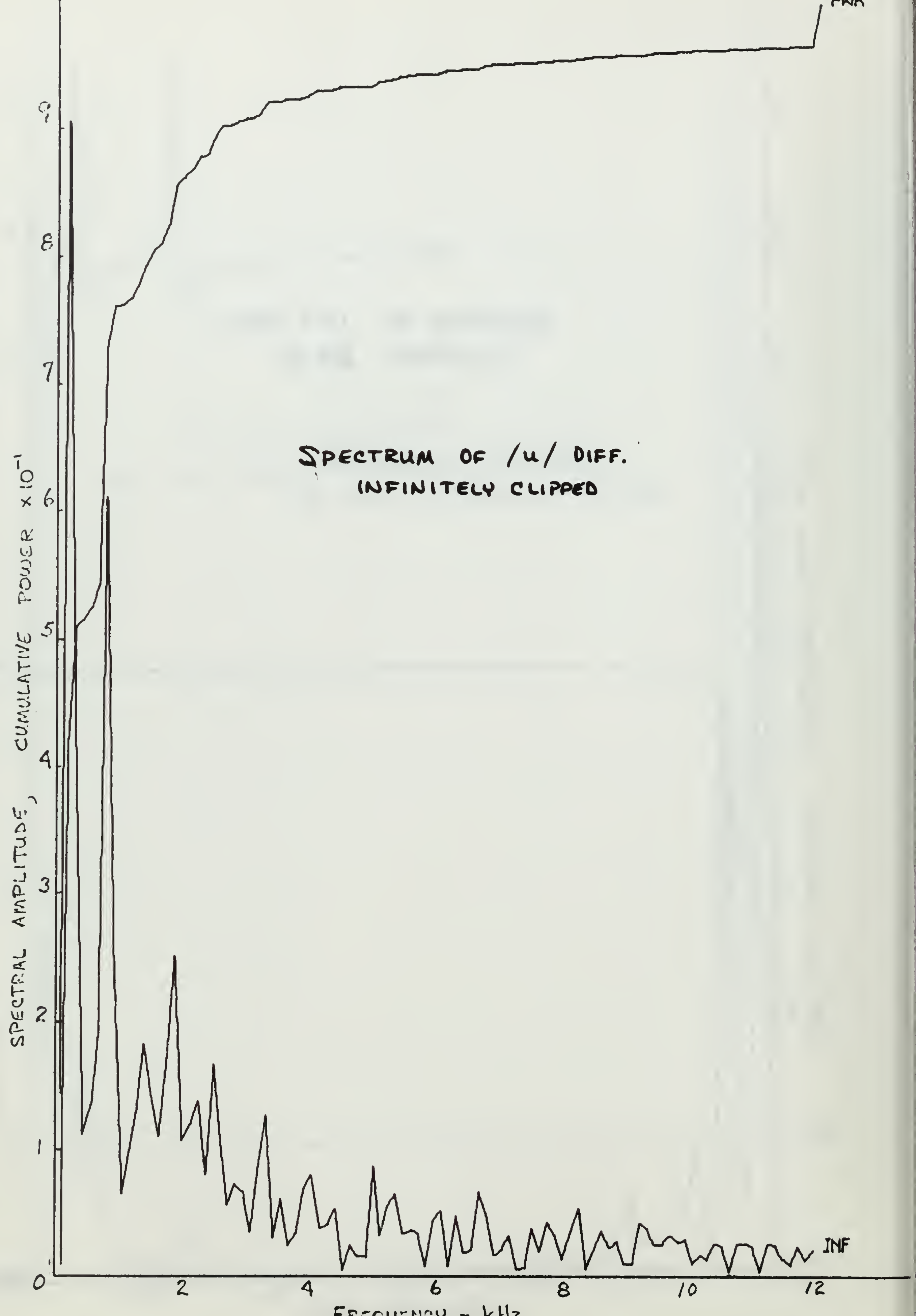

SPECTRUM OF /u/
INFINITELY CLIPPED
PWR
INF
SPECTRAL AMPLITUDE, CUMULATIVE POWER x 10^{-1}
FREQUENCY - kHz
0
2
4
6
8
10
12
1
2
3
4
5
6
7
8
9



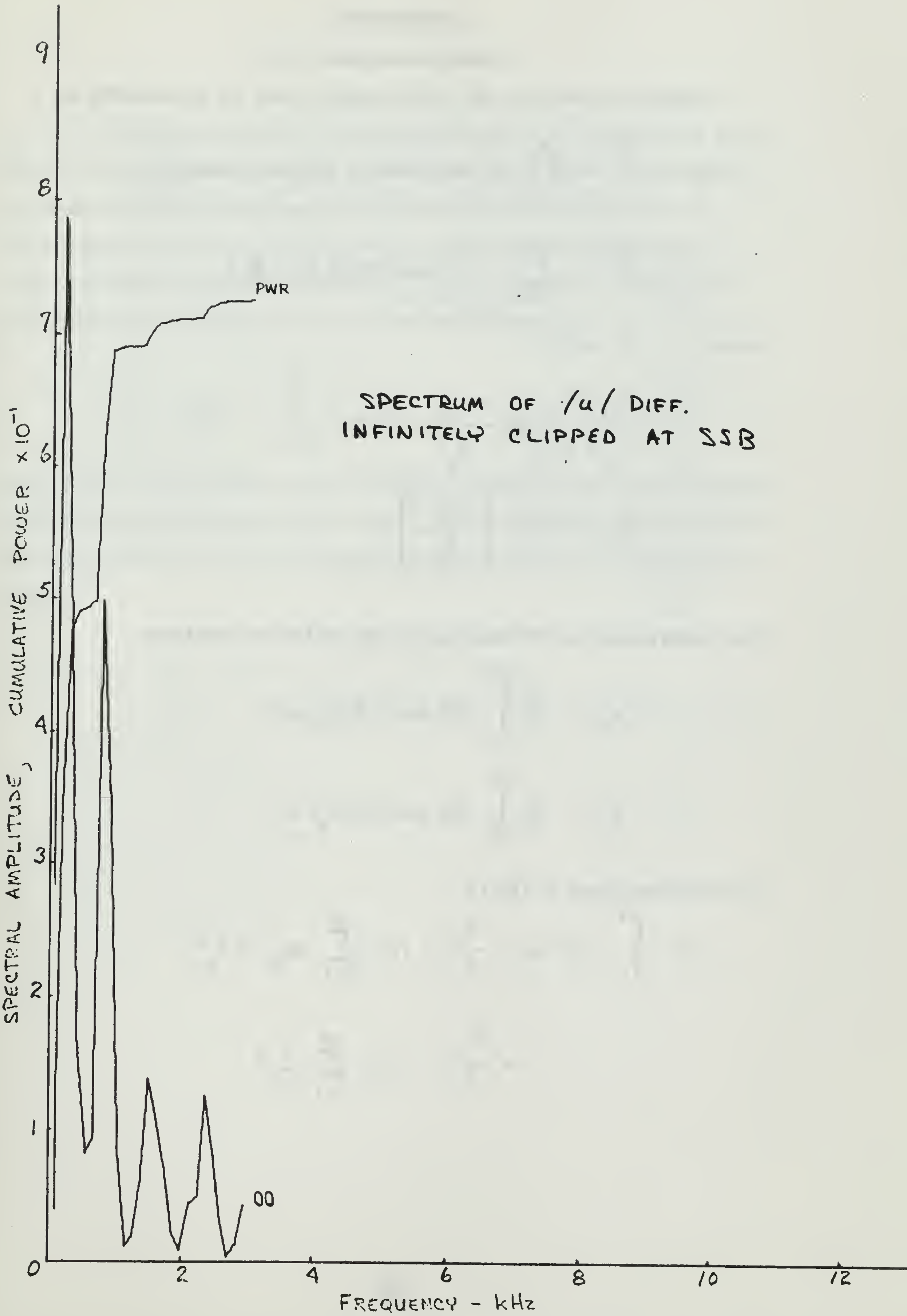
SPECTRUM OF /u/
INFINITELY CLIPPED AT SSB
PWR
SPECTRAL AMPLITUDE, CUMULATIVE POWER × 10⁻¹
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


ORIGINAL SPECTRUM
for /u/ DIFFERENTIATED
PHASE - rad.
1
0
-1
PHI
AMPLITUDE
4
3
2
1
0
PWR
.00
2
4
6
8
10
12
FREQUENCY - kHz


SPECTRUM OF /u/ DIFF.
CLIPPED 10 db
SPECTRAL AMPLITUDE, CUMULATIVE POWER x 10^-1
PWR
FREQUENCY - kHz
1008



SPECTRUM OF /u/ DIFF.
INFINITELY CLIPPED
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^-1
FREQUENCY - kHz
INF
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12


SPECTRUM OF /u/ DIFF.
INFINITELY CLIPPED AT SSB
PWR
00
SPECTRAL AMPLITUDE, CUMULATIVE POWER x10^-1
FREQUENCY - kHz
0
1
2
3
4
5
6
7
8
9
2
4
6
8
10
12

APPENDIX B

Fourier Analysis

A periodic function, f(t), with period T may be represented as

$$f(t) = \frac{A_o}{2} + \sum_{n=1}^{\infty} (A_n \cos 2\pi n f_o t + B_n \sin 2\pi n f_o t)$$

$$= \frac{C_o}{2} + \sum_{n=1}^{\infty} C_n \cos (2\pi n f_o t + \phi_n)$$

where $f_o = 1/T$

$C_o = A_o$

$C_n = \sqrt{A_n^2 + B_n^2}$

$\phi_n = \tan^{-1} \left[\frac{-B_n}{A_n} \right]$

The coefficients are determined by the following relations

$$A_n = \frac{2}{T} \int_o^T f(t) \cos 2\pi n f_o t \, dt$$

$$B_n = \frac{2}{T} \int_o^T f(t) \sin 2\pi n f_o t \, dt$$

The average power in f(t) is

$$1/T \int_o^T f(t)^2 \, dt = \frac{A_o}{4} + 1/2 \sum_{n=1}^{\infty} (A_n^2 + B_n^2)$$

$$= \frac{C_o}{4} + 1/2 \sum_{n=1}^{\infty} C_n^2$$

APPENDIX C

Digital Computer Programs

The calculations for this study were made by a Control Data 1604 digital computer with a Fortran 63 compiler. The major portion of the computations involved transformations back and forth between the time domain and the frequency domain. The transformations were accomplished using the principles of Fourier analysis. The integrals involved were evaluated by making the approximation

$$\int_{o}^{T} f(x)dx \simeq \sum_{i=1}^{N} f(i\,\Delta x)\,\Delta x \qquad \text{where} \qquad \Delta x = \frac{T}{N}$$

Values of N were chosen so that the errors made by this approximation did not significantly affect the results. Listed on the following pages are the subroutines used in making the calculations discussed in this study.

```
      SUBROUTINE ANALYZE (A,B,C,FREQ,LIM)
C           PERFORMS FOURIER ANALYSIS OF A WAVEFORM IN ARRAY S
C           CONSIDERED AS A FUNCTION OF TIME (ARRAY T).  A AND
C           B ARE OUTPUT ARRAYS OF FOURIER COEFFICIENTS AND
C           C = A**2 + B**2.  FREQ IS THE FUNDAMENTAL
C           FREQUENCY.  LIM IS THE NUMBER OF POINTS IN ARRAYS
C           S AND T.
      DIMENSION A(100),B(100),C(100),S(900),T(900)
      COMMON/B1/S/B2/T
      ARG2=2.*3.1415926536*FREQ
      FORM=2./FLOATF(LIM)
   29 DO 30 N=1,100
      AN=0.
      BN=0.
      ARG1=ARG2*FLOATF(N)
      DO 35 I=1,LIM
      ARG=ARG1*T(I)
      A N =A N +S(I)*COSF(ARG)
   35 B N =B N +S(I)*SINF(ARG)
      A(N)=A N *FORM
      B(N)=B N *FORM
   30 C(N)=     (A(N)**2+B(N)**2)
      RETURN
      END
```

```
      SUBROUTINE AN (A,B,C)
C          PERFORMS FOURIER ANALYSIS OF AN INFINITELY CLIPPED
C          WAVEFORM IN ARRAY S BY SUPERPOSITION OF THE
C          SPECTRA OF THE COMPONENT PULSES.  A AND B ARE THE
C          OUTPUT ARRAYS OF FOURIER COEFFICIENTS AND
C          C = A**2 + B**2.
      DIMENSION A(100),B(100),AI(100),BI(100),Z(100),C(100)
      CALL ZEROS (900,Z)
      DO 10 I=1,100
   10 A(I)=B(I)=0.
      DO 22 I=1,100,2
      IF (Z(I)) 21,23,21
   21 TO=Z(I+1)-Z(I)
      T1=Z(I)+TO/2.
      CALL SPECTRUM (-20.,1./120.,TO,T1,AI,BI)
      DO 22 N=1,100
      A(N)=A(N)+AI(N)
   22 B(N)=B(N)+BI(N)
   23 DO 24 I=1,100
   24 C(I)=A(I)**2+B(I)**2
      RETURN
      END


      SUBROUTINE SPECTRUM (AMP,T,TO,T1,A,B)
C          COMPUTES THE SPECTRUM OF A PULSE OF HEIGHT AMP
C          AND WIDTH `TO' CENTERED AT TIME T1. T IS THE PERIOD.
      DIMENSION A(100),B(100)
      CO=AMP*TO/T*2.
      ARG1=3.1415926536*TO/T
      ARG2=3.1415926536*2.*T1/T
      DO 20 N=1,100
      ARG=ARG1*FLOATF(N)
      C=CO*SINF(ARG)/ARG
      PHI=-ARG2*FLOATF(N)
      A(N)=C*COSF(PHI)
   20 B(N)=-C*SINF(PHI)
      RETURN
      END
```

```
      SUBROUTINE ZEROS (NUM,Z)
C           STORES THE TIMES OBTAINED FROM ARRAY T  OF ZERO
C           CROSSING OF THE WAVEFORM IN ARRAY S (NUMBER OF
C           POINTS = NUM)  INTO THE ARRAY Z.  AFTER THE LAST
C           ZERO CROSSING HAS BEEN ENTERED, SUCCEEDING VALUES
C           OF Z ARE EQUAL TO ZERO.
      DIMENSION S(900),T(900),Z(100)
      COMMON/B1/S/B2/T
      J=1
      LAST=1
      DO 10 I=1,100
   10 Z(I)=0.
      DO 25 I=1,NUM
      IF (S(I)) 21,22,22
   21 NOW=-1
      GO TO 23
   22 NOW=1
   23 IF(NOW+LAST) 25,24,25
   24 Z(J)=T(I)
      J=J+1
   25 LAST=NOW
      RETURN
      END
```

```
      SUBROUTINE SYN (A,B,FREQ,ITYPE,NUM)
C           PERFORMS FOURIER SYNTHESIS.  IF ITYPE IS ZERO,
C           A AND B ARE INTERPRETED AS THE CONVENTIONAL FOURIER
C           COEFFICIENTS.  IF ITYPE IS NON-ZERO, A IS
C           INTERPRETED AS THE C COEFFICIENT AND B AS THE
C           PHASE ANGLE IN RADIANS.  NUM IS THE NUMBER OF POINTS
C           TO BE USED OF THE A AND B ARRAYS.  FREQ IS THE
C           FUNDAMENTAL FREQUENCY.
      DIMENSION S(900),T(900),A(50),B(50)
      COMMON/B1/S/B2/T
      DO 21 I=1,900
      S(I)=0.
      ARG1=2.*3.1415926536*FREQ*T(I)
      DO 20 N=1,NUM
      ARG=ARG1*FLOATF(N)
      IF (ITYPE) 15,10,15
   10 S(I)=S(I)+A(N)*COSF(ARG)+B(N)*SINF(ARG)
      GO TO 20
   15 S(I)=S(I)+A(N)*COSF(ARG+B(N))
   20 CONTINUE
   21 CONTINUE
      RETURN
      END


      SUBROUTINE NORM (NUM)
C           NORMALIZES THE ARRAY S (CONTAINING NUM POINTS) SO
C           THAT THE MAXIMUM POSITIVE VALUE IS +10.
      DIMENSION S(900)
      COMMON/B1/S
      SMAX=0.
      DO 21 I=1,NUM
      IF(SMAX-S(I))22,21,21
   22 SMAX=S(I)
   21 CONTINUE
      GAIN=10./SMAX
      DO 23 I=1,NUM
   23 S(I)=S(I)*GAIN
      RETURN
      END
```

```
      SUBROUTINE CLIP (CLPLVL)
C         AMPLIFIES THE ARRAY S BY THE FACTOR CLPLVL AND
C         CLIPS SYMMETRICALLY AT A VALUE OF 10.
      DIMENSION S(900)
      COMMON/B1/S
   28 DO 27 I=1,900
      S(I)=S(I)*CLPLVL
      IF (S(I)-10.) 24,26,26
   24 IF (S(I)+10.) 25,27,27
   25 S(I)=-10.
      GO TO 27
   26 S(I)=10.
   27 CONTINUE
      RETURN
      END


      SUBROUTINE GRADCLIP (S,NUM,CLPLVL)
C         CLIPS THE WAVEFORM IN ARRAY S ACCORDING TO THE
C         CHARACTERISTIC  Y = 10 * TANH( X / 10 ).
      DIMENSION S(10000)
      DO 10 I=1,NUM
   10 S(I)=10.*TANHF(S(I)*CLPLVL/10.)
      RETURN
      END


      SUBROUTINE POWER (PWR,LIM)
C         COMPUTES THE AVERAGE POWER IN THE ARRAY S.
      DIMENSION S(900)
      COMMON/B1/S
      PWR=0.
      DO 30 I=1,LIM
   30 PWR=PWR+S(I)**2
      PWR=PWR/FLOATF(LIM)
      RETURN
      END
```

```
      SUBROUTINE POWERF (C,PF)
C        COMPUTES THE CUMULATIVE DISTRIBUTION FUNCTION
C        OF THE POWER IN THE ARRAY C.
      DIMENSION C(100),PF(100)
      PF(1)=C(1)/2.
      DO 10 I=2,100
   10 PF(I)=PF(I-1)+C(I)/2.
      RETURN
      END


      SUBROUTINE COHERE (C1,C2,NUM,COEF)
C        COMPUTES THE COHERENCE COEFFICIENT BETWEEN NUM
C        POINTS OF THE ARRAYS C1 AND C2.
      DIMENSION C1(100),C2(100)
      SUM=0.
      SIGSQ1=0.
      SIGSQ2=0.
      DO 10 I=1,NUM
      SIGSQ1=SIGSQ1+C1(I)**2
      SIGSQ2=SIGSQ2+C2(I)**2
   10 SUM=SUM+(C1(I)*C2(I))
      COEF=SUM/ SQRTF(SIGSQ1*SIGSQ2)
      RETURN
      END
```

```
      SUBROUTINE SMOOTH (NUM,SINP,LIM)
C           INTERPOLATES A SMOOTH CURVE BETWEEN NUM INPUT
C           POINTS IN ARRAY C BY MEANS OF SIN(X)/X FUNCTION.
C           OUTPUT IS THE ARRAY S AND WILL HAVE LIM POINTS.
      DIMENSION S(900),T(900),SINP(100)
      COMMON/B1/S/B2/T
      INT=900/(NUM-1)
      LIM=INT*(NUM-1)
      SAVG=0.
      DO 10 I=1,NUM
   10 SAVG=SAVG+SINP(I)
      SAVG=SAVG/FLOATF(NUM)
      DO 11 I=1,NUM
   11 SINP(I)=SINP(I)-SAVG
      DO 20 I=1,900
      T(I)=FLOATF(I)/FLOATF(LIM)/120.
   20 S(I)=0.
      DO 30 J=1,NUM
      DO 30 I=1,LIM
      ARG=3.1415926536*FLOATF (I-(J-1)*INT)/FLOATF(INT)+1.0E-50
   30 S(I)=S(I)+SINP(J)*SINF(ARG)/ARG
      RETURN
      END
```

```
      SUBROUTINE PHASE (NUM, C, PHI)
C            INFERS A PHASE CHARACTERISTIC BASED ON THE
C            AMPLITUDE FUNCTION IN ARRAY C.  PHI IS THE OUTPUT
C            PHASE ARRAY IN RADIANS.
      DIMENSION S(900),T(900),PHI1(600),C(50),PHI(50),MAX(50),X(300),
     *   Y(300)
      COMMON/B1/S/B2/T
      ILAST=NUM*12
      DO 70 I=1,NUM
      X(I)=120.*FLOATF(I)
   70 Y(I)=C(I)
      DO 71 I=12,ILAST
      XINT=10.*FLOATF(I)
      CALL SPLINE (X,Y,NUM,XINT,YINT)
   71 S(I)=YINT
      IPOS=1
      J=1
      DO 50 I=12,ILAST
      IF (S(I+1)-S(I))46,49,49
   46 IF (IPOS) 50,50,47
   47 MAX(J)=I
      J=J+1
      IPOS=0
      GO TO 50
   49 IPOS=1
   50 CONTINUE
      JLAST=J-1
      DO 55 I=1,600
   55 PHI1(I)=0.
      DO 60 J=1,JLAST
      FMAXJ=MAX(J)
      Q=FMAXJ/10.
      DO 60 I=1,600
      FI=I
   60 PHI1(I)=ATANF(1./(Q*(FMAXJ /FI-FI/FMAXJ )+1.0E-20))  +PHI1(I)
      DO 65 I=1,50
   65 PHI(I)=PHI1(12*I)
      RETURN
      END
      SUBROUTINE SPLINE(X,Y,M,XINT,YINT)
      DIMENSION X(300),Y(300),C(4,300)
      IF(X(1)+Y(M)-ATER)10,3,10
   10 CALL SPLICON(X,Y,M,C)
      ATER=X(1)+Y(M)
      K=1
    3 IF(XINT-X(1)) 70,1,2
   70 K=1
```

```
      GO TO 7
    1 YINT=Y(1)
      RETURN
    2 IF(XINT-X(K+1))6,4,5
    4 YINT=Y(K+1)
      RETURN
    5 K=K+1
      IF(M-K) 71,71,3
   71 K=M-1
      GO TO 7
    6 IF(XINT-X(K))13,12,11
   12 YINT=Y(K)
      RETURN
   13 K=K-1
      GO TO 6
   11 YINT=(X(K+1)-XINT)*(C(1,K)*(X(K+1)-XINT)**2+C(3,K))
      YINT=YINT+(XINT-X(K))*(C(2,K)*(XINT-X(K))**2+C(4,K))
      RETURN
    7 PRINT 101,XINT
  101 FORMAT(8H0XINT = E18.9,32H, OUT OF RANGE FOR INTERPOLATION)
      GO TO 11
      END
      SUBROUTINE SPLICON(X,Y,M,C)
      DIMENSION X(300),Y(300),C(4,300),D(300),P(300),E(300),A(300,3),B(3
     100),Z(300)
      MM=M-1
      DO 2 K=1,MM
      D(K)=X(K+1)-X(K)
      P(K)=D(K)/6.
    2 E(K)=(Y(K+1)-Y(K))/D(K)
      DO 3 K=2,MM
    3 B(K)=E(K)-E(K-1)
      A(1,2)=-1.-D(1)/D(2)
      A(1,3)=D(1)/D(2)
      A(2,3)=P(2)-P(1)*A(1,3)
      A(2,2)=2.*(P(1)+P(2))-P(1)*A(1,2)
      A(2,3)=A(2,3)/A(2,2)
      B(2)=B(2)/A(2,2)
      DO 4 K=3,MM
      A(K,2)=2.*(P(K-1)+P(K))-P(K-1)*A(K-1,3)
      B(K)=B(K)-P(K-1)*B(K-1)
      A(K,3)=P(K)/A(K,2)
    4 B(K)=B(K)/A(K,2)
      Q=D(M-2)/D(M-1)
      A(M,1)=1.+Q+A(M-2,3)
      A(M,2)=-Q-A(M,1)*A(M-1,3)
      B(M)=B(M-2)-A(M,1)*B(M-1)
      Z(M)=B(M)/A(M,2)
      MN=M-2
      DO 6 I=1,MN
```

```
      K=M-I
    6 Z(K)=B(K)-A(K,3)*Z(K+1)
      Z(1)=-A(1,2)*Z(2)-A(1,3)*Z(3)
      DO 7 K=1,MM
      Q=1./(6.*D(K))
      C(1,K)=Z(K)*Q
      C(2,K)=Z(K+1)*Q
      C(3,K)=Y(K)/D(K)-Z(K)*P(K)
    7 C(4,K)=Y(K+1)/D(K)-Z(K+1)*P(K)
      END
```

```
C          THE FOLLOWING PROGRAM ILLUSTRATES THE USAGE OF
C          SOME OF THE SUBROUTINES.  IT ALSO CONTAINS THE
C          STATEMENTS NECESSARY TO GENERATE A SSB SIGNAL AND
C          ANALYZE IT.
      PROGRAM CLIPSPEC
      DIMENSION S(900),T(900),SN(10000),A(100),B(100),C(100),IT(12),
     *   ORD(900),ABS(900),CA(100),PHI(50),C1(100),LAB(10)
      COMMON/B1/S/B2/T
      LAB(1)=4H10DB   $   LAB(2)=4H20DB   $   LAB(3)=4H INF
      DO 4 I=1,50
    4 PHI(I)=0.
      DO 5 I=1,100
    5 A(I)=B(I)=C(I)=C1(I)=0.
      READ 9, LA, NUM
    9 FORMAT (A4,I3)
      READ 12, (C(I),I=1,NUM)
   12 FORMAT (20F4.1)
      DO 10 I=1,NUM
   10 C(I)=EXPF(C(I)*.230258509/2.)
C          INFER PHASE CHARACTERISTIC
      CALL PHASE (NUM,C,PHI)
      DO 6 I=1,900
    6 T(I)=FLOATF(I)/900./120.
      CALL SYN (C,PHI,120.,1)
      SFIRST=S(1)
      CALL NORM (900)
      GAIN=S(1)/SFIRST
      DO 7 I=1,NUM
    7 C(I)=C(I)*GAIN
      DO 8 I=1,NUM
    8 C1(I)=C(I)**2
      CALL POWERF(C1,A)
      CALL POWER (PWR,900)
      PRINT 100,PWR,A(100)
  100 FORMAT(1H1,9X14HORIGINAL POWER,2F20.3///)
      DO 13 I=1,100
      ORD(I)=C(I)
   13 ABS(I)=120.*FLOATF(I)
      CALL ITITLE (IT)
      IT(1)=8HORIGINAL
      IT(2)=8H AUDIO
      IT(3)=8HSPECTRUM
      CALL DRAW(100,ABS,ORD,1,0,LA,IT,2000.,1.,0,0,2,2,7,10,0,L)
      DO 135 I=1,50
  135 ORD(I)=PHI(I)+8.
      CALL DRAW (50,ABS,ORD,2,0,4H PHI,2000.,1.,0,0,2,2,7,10,0,L)
      DO 14 I=1,100
   14 ORD(I)=A(I)/10.
      CALL DRAW(100,ABS,ORD,3,0,4H PWR,IT,2000.,1.,0,0,2,2,7,10,0,L)
      IT(1)=8HCLIPPED
```

```
      CLPLVL=3.16
      DO 20 J=1,3
   16 CALL GRADCLIP (S,900,CLPLVL)
      CALL ANALYZF (A,B,CA,120.,900)
      CALL POWERF (CA,A)
      CALL POWER (PWR,900)
      A(100)=PWR
      CALL COHERE (C1,CA,NUM,COEF)
      PRINT 101,A(NUM),PWR,COEF
  101 FORMAT(10X,17HPOWER IN  SIGNAL ,F20.3/10X11HTOTAL POWER,F20.3/
     *10X21HCOHERENCE COEFFICIENT,F20.10///)
      DO 17 I=1,100
   17 ORD(I)=SQRTF(CA(I))
      CALL DRAW(100,ABS,ORD,1,0,LAB(J),IT,2000.,1.,0,0,2,2,7,10,0,L)
      DO 18 I=1,100
   18 ORD(I)=A(I)/10.
      CALL DRAW(100,ABS,ORD,3,0,4H PWR,IT,2000.,1.,0,0,2,2,7,10,0,L)
   20 CONTINUE
      TWOPI=6.2831853072
      ARG2=TWOPI/1200000.
      DO 30 I=1,10000
      SN(I)=0.
      ARG1=ARG2*FLOATF(I)
      DO 30 N=1,NUM
      ARG=ARG1*(24000.+120.*FLOATF(N))
   30 SN(I)=SN(I)+C(N)*COSF(ARG+PHI(N))
      CALL GRADCLIP (SN,10000,10.)
      DO 51 N=1,NUM
      A(N)=0.
      B(N)=0.
      ARG1=TWOPI*(24000.+120.*FLOATF(N))/1200000.
      DO 50 I=1,10000
      ARG=ARG1*FLOATF (I)
      A(N)=A(N)+SN(I)*COSF(ARG)
   50 B(N)=B(N)+SN(I)*SINF(ARG)
      A(N)=A(N)/5000.
      B(N)=B(N)/5000.
      CA(N)  =    (A(N)**2+B(N)**2)
   51 ORD(N)=SQRTF(CA(N))
      IT (2)=8H   SSB
      CALL DRAW(NUM,ABS,ORD,1,0,LA,IT,2000.,1.,0,0,2,2,7,10,0,L)
      CALL POWERF (CA,A)
      CALL COHERE (C1,CA,NUM,COEF)
      PRINT 102, A(NUM),COEF
  102 FORMAT(10X9HSSB POWER,F20.3/10X21HCOHERENCE COEFFICIENT,F20.10//)
      DO 52 I=1,100
   52 ORD(I)=A(I)/10.
      CALL DRAW(100,ABS,ORD,3,0,4H PWR,IT,2000.,1.,0,0,2,2,7,10,0,L)
   55 END
```

INITIAL DISTRIBUTION LIST

		No. Copies
1.	Defense Documentation Center Cameron Station Alexandria, Virginia 22314	20
2.	Library Naval Postgraduate School Monterey, California 93940	2
3.	Commander, Naval Ship Systems Command Department of the Navy Washington, D. C. 20360	1
4.	Commander, Naval Electronic Systems Command Department of the Navy Washington, D. C. 20360	1
5.	Dr. Gerald D. Ewing Department of Electrical Engineering Naval Postgraduate School Monterey, California 93940	3
6.	LT William V. Dietrich, USN c/o Naval Destroyer School Naval Base Newport, Rhode Island	1

UNCLASSIFIED
Security Classification

DOCUMENT CONTROL DATA - R&D

(Security classification of title, body of abstract and indexing annotation must be entered when the overall report is classified)

1. ORIGINATING ACTIVITY *(Corporate author)*
Naval Postgraduate School
Monterey, California 93940

2a. REPORT SECURITY CLASSIFICATION
UNCLASSIFIED

2b. GROUP

3. REPORT TITLE

Calculations of the Effects of Peak Clipping on Speech-like Signals

4. DESCRIPTIVE NOTES *(Type of report and inclusive dates)*
Master's Thesis - December 1966

5. AUTHOR(S) *(Last name, first name, initial)*

Dietrich, William V. LT United States Navy

6. REPORT DATE	7a. TOTAL NO. OF PAGES	7b. NO. OF REFS
December 1966	128	26

8a. CONTRACT OR GRANT NO.

b. PROJECT NO.

c.

d.

9a. ORIGINATOR'S REPORT NUMBER(S)

9b. OTHER REPORT NO(S) *(Any other numbers that may be assigned this report)*

[illegible] 12/16/69

10. AVAILABILITY/LIMITATION NOTICES

This document has been approved [illegible] release and sale; its distribution is unlimited.

[redacted].

11. SUPPLEMENTARY NOTES

12. SPONSORING MILITARY ACTIVITY

13. ABSTRACT

Peak clipping is a well known method of increasing the average power output of a peak power limited voice communication transmitter. Although the clipping process introduces distortion, articulation tests have shown that clipped speech remains highly intelligible. Using idealizations of vowel sounds based on the mechanism of speech production, calculations were made of the spectra resulting from clipping these speech-like signals. The results indicate a high degree of similarity between the spectra before and after clipping. The power gained by clipping at audio frequency and at narrowband was calculated and compared with previously published data. Repeaking due to component rejection was investigated for clipping at audio and narrowband. Calculations of the effect of varying the phase characteristic of the signals before clipping indicate that such variation may improve the intelligibility of clipped speech.

DD FORM 1473 1 JAN 64

UNCLASSIFIED
Security Classification

14. KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Speech Speech models Clipping Repeaking Speech spectra Distortion						

INSTRUCTIONS

1. ORIGINATING ACTIVITY: Enter the name and address of the contractor, subcontractor, grantee, Department of Defense activity or other organization *(corporate author)* issuing the report.

2*a*. REPORT SECURITY CLASSIFICATION: Enter the overall security classification of the report. Indicate whether "Restricted Data" is included. Marking is to be in accordance with appropriate security regulations.

2*b*. GROUP: Automatic downgrading is specified in DoD Directive 5200.10 and Armed Forces Industrial Manual. Enter the group number. Also, when applicable, show that optional markings have been used for Group 3 and Group 4 as authorized.

3. REPORT TITLE: Enter the complete report title in all capital letters. Titles in all cases should be unclassified. If a meaningful title cannot be selected without classification, show title classification in all capitals in parenthesis immediately following the title.

4. DESCRIPTIVE NOTES: If appropriate, enter the type of report, e.g., interim, progress, summary, annual, or final. Give the inclusive dates when a specific reporting period is covered.

5. AUTHOR(S): Enter the name(s) of author(s) as shown on or in the report. Enter last name, first name, middle initial. If military, show rank and branch of service. The name of the principal author is an absolute minimum requirement.

6. REPORT DATE: Enter the date of the report as day, month, year; or month, year. If more than one date appears on the report, use date of publication.

7*a*. TOTAL NUMBER OF PAGES: The total page count should follow normal pagination procedures, i.e., enter the number of pages containing information.

7*b*. NUMBER OF REFERENCES: Enter the total number of references cited in the report.

8*a*. CONTRACT OR GRANT NUMBER: If appropriate, enter the applicable number of the contract or grant under which the report was written.

8*b*, 8c, & 8*d*. PROJECT NUMBER: Enter the appropriate military department identification, such as project number, subproject number, system numbers, task number, etc.

9*a*. ORIGINATOR'S REPORT NUMBER(S): Enter the official report number by which the document will be identified and controlled by the originating activity. This number must be unique to this report.

9*b*. OTHER REPORT NUMBER(S): If the report has been assigned any other report numbers *(either by the originator or by the sponsor)*, also enter this number(s).

10. AVAILABILITY/LIMITATION NOTICES: Enter any limitations on further dissemination of the report, other than those imposed by security classification, using standard statements such as:

(1) "Qualified requesters may obtain copies of this report from DDC."

(2) "Foreign announcement and dissemination of this report by DDC is not authorized."

(3) "U. S. Government agencies may obtain copies of this report directly from DDC. Other qualified DDC users shall request through
____."

(4) "U. S. military agencies may obtain copies of this report directly from DDC. Other qualified users shall request through
____."

(5) "All distribution of this report is controlled. Qualified DDC users shall request through
____."

If the report has been furnished to the Office of Technical Services, Department of Commerce, for sale to the public, indicate this fact and enter the price, if known.

11. SUPPLEMENTARY NOTES: Use for additional explanatory notes.

12. SPONSORING MILITARY ACTIVITY: Enter the name of the departmental project office or laboratory sponsoring *(paying for)* the research and development. Include address.

13. ABSTRACT: Enter an abstract giving a brief and factual summary of the document indicative of the report, even though it may also appear elsewhere in the body of the technical report. If additional space is required, a continuation sheet shall be attached.

It is highly desirable that the abstract of classified reports be unclassified. Each paragraph of the abstract shall end with an indication of the military security classification of the information in the paragraph, represented as *(TS)*, *(S)*, *(C)*, *or (U)*.

There is no limitation on the length of the abstract. However, the suggested length is from 150 to 225 words.

14. KEY WORDS: Key words are technically meaningful terms or short phrases that characterize a report and may be used as index entries for cataloging the report. Key words must be selected so that no security classification is required. Identifiers, such as equipment model designation, trade name, military project code name, geographic location, may be used as key words but will be followed by an indication of technical context. The assignment of links, roles, and weights is optional.